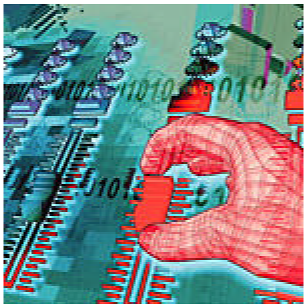
Digital Mixing



It's increasingly popular for both the home and pro studios

If you take a look through the trade ads these days, you'll notice digital mixers take pride of place over any analogue desks which happen to be in production. But what has happened to the business of mixing to make everyone want a digital mixer instead of an analogue one? It wasn't long ago when you could judge a studio's position in the market by looking at the desk. If it was at least six -foot long with a patchbay the size of a filing cabinet, it must be a pretty serious studio.

These days, some of the best studios have desks no bigger than a coffee table and home studios often feature similar desks. Digital mixers have changed the way we work and are changing the way we view the traditional recording studio.

Total recall

Ten years ago, only desks like the revered SSL had any kind of recall facility. Crude by today's standards, it still involved putting each and every pot back to the required position yourself, even if the computer did tell you where that was. Back in the 80s, Yamaha's DMP7 digital mixer offered eight channels of digital mixing with internal effects and motorised faders for a lot less than most equivalent analogue systems, but it was the early 90s before things really took off with the Yamaha Pro Mix 01 (or Programmable Mixer 01, as it was later renamed for copyright reasons). This featured 16 channels with onboard effects, motorised faders and a decent MIDI spec. Using snapshot memories to recall a mix was suddenly a possibility. At this point, several other manufacturers entered the market and soon just about every R&D department in the business was working on an affordable digital mixer.

Over the following years, several models became the main players for the project and pro studio alike, namely the Yamaha 02R, the Spirit 328 and the Mackie D8B. There are plenty of others, especially in the high-end market, but these tend to have the size and price of large analogue desks, so they're not really an option for most of us.

The features we've come to expect from digital desks include full EQ on every channel, dynamics on every channel, internal effects, total recall via snapshots and automation either via a MIDI sequencer or internal. It's a far cry from a 24-input analogue desk with limited EQ on the monitor section, no effects, no dynamic control and no recall other than mutes and possibly fader positions, but that's how far we've come. There will always be a place for analogue desks, but that place is becoming harder to find every day, especially with the prices of the new breed of digital mixers.

Affordability

When it comes to working in pro studios, the desk you use depends on who booked the studio and what you've got to record but, for example, for recording strings an analogue Neve desk would be our first choice for its warmth and good mic amps. Most of the larger studios big enough to record strings are fitted with one. If these really big studios do have a digital desk, it will usually be the likes of the Euphonix or, on rare occasions, the Sony Oxford. The kind of desk we're talking about here however, is the kind we can actually afford, like the Spirit 328 and the Mackie D8B.

The 328 and the D8B are very similar. Unlike the Yamaha range and others, they have more of the feel of an analogue desk. This has earned them many admirers because, being similar to analogue desks, they are more intuitive to learn.

The one big difference between the two is the Mackie is a lot more expensive. Both are great to record with, both use 24-bit converters so quality is high and they both have great EQ. The 328 and D8B use rotary encoders with a segment display that shows the value of a pot. Both use fader switching so you have a different bank of faders at the press of a switch. This means the desks can perform as tape returns, group sends, even MIDI controllers.

This kind of versatile instant switching is one of the best things about these desks, not forgetting the other big

advantage: snapshot store and recall. Restoring all EQ and effects settings on analogue desks involved either lining up the pots back to where they were on a screen, or pages and pages of written settings for all the outboard gear. By using more plug-ins and internal effects built into the digital desks, the need for outboard gear is reduced and switching between songs takes five minutes rather than two hours to get the balance of the mix back. For songwriting this is a fantastic way to switch between songs rather than having to stay on the one song because you don't want to change the desk now you have finally got it sounding good.

Professional sound

People sometimes believe they can't get professional results in their home studios because of their desk, but with these types of digital desks you can. The digital EQ is different and ideally you still need the acoustic environment of a good studio to achieve a great sound, but it's possible as one of Alan Branch's recent projects proves.

"Recently, I got involved in writing with a band from Leeds called LSK. I used two Spirit 328s linked up, which gave me a total of 84 inputs, not including the tape returns! We wrote, recorded and mixed loads of tracks in a little farmhouse using Logic, some plug-ins and the Lexicon effects built into the 328 desks. The album is out now, has had fantastic reviews and topped the charts in Japan, with singles outselling Madonna!"

You may think it's easy for us to say as engineer/producers, but it is possible to get great results from digital desks like these. And with the Recall function there is little excuse to keep you from going back to perfect the mix.

Beware of the hype you read about digital desks. We both remember a few articles by certain non-studio working journalists about how important it is to see the EQ curve and how a mix seemed weird without moving faders (motors off). This is nonsense. Do you really think professional engineers who mix records every day use a screen to see the EQ? No, they use their ears.

A screen might actually put your hearing off, as you're using one of your senses to look at the EQ rather than listening. And as for moving faders, engineers and activists used to joke they were only good for the A&R guy to look at. But seriously, motorised faders can be useful to see what's going on, but not at the price of not listening to the mix. A good tip when programming is to look away from the screen when trying to concentrate on a certain sound, such as checking a crossfade.

How they work

To understand the advantages of a digital mixer, you need to know the principles behind their working methods. In basic terms, once an audio source enters the desk either via an A/D (analogue to digital) converter or digitally via one of the many digital formats, everything is done in the digital domain. Zeros and ones are the language of digital gear and no matter what you do to the signal, all that is really happening is the information is being modified. Due to this process, no noise, crosstalk, distortion or unwanted degradation of the signal can occur.

Obviously, all this depends on the bit depth of the internal processing and more importantly, the quality of the A/D and D/A converters. With more and more studios using digital multitracks or DAWs, the connections between the desk and 'recorder' are often also digital. All this helps keep noise out and sonic integrity to the fore.

As anyone who uses an analogue desk will know, switching between projects is a time-draining and tedious exercise involving making notes of settings,
chinagraphing channel strips and storing effects settings on all your outboard. I've even seen engineers take Polaroids of the desk to help assist with resetting the mix later. With snapshot recall, all that is a thing of the past. Just hit the Store button and name the file. With the press of a button you can recall absolutely everything on the desk including levels, EQ, effects, compression, gating, expansion, routing and synchronisation settings. As most project studios revolve around MIDI set-ups, you can easily recall your entire studio's 'status' with a digital mixer and a SysEx message from your sequencer.

The advantages of recall for a small studio are enormous. You can work on many projects at the same time and switch between them. If the band you recorded last week wants to come and do a mix half an hour after you finish another tracking session, you can do it. In a writing studio, if you're recording vocals on one track, then you suddenly decide you have a great idea for another track you're working on, no problem. It's easy with recall. Anyone working with sound to picture projects for various clients can instantly recall a mix when they drop in to see how it's going. It makes life easy when the client wants that snare just a touch louder than when you mixed the track a week ago and have used the desk for something else since.

Another obvious advantage is price. For under five grand you can get a 40-channel desk with two effects processors and proper EQ with all the other advantages of digital. Try getting an analogue desk with that spec,

then add the cost of all those compressors, gates, effects and an automation system. No chance. With so many big name producers using these desks for some of the biggest records you've heard lately, there's obviously not a problem with quality either, so it's almost daft not to go digital.

The last advantage is the size. The footprint of most digital desks is tiny compared to an analogue equivalent. This means you really can have a pro studio at home. You don't need an enormous desk to get a pro quality sound. Also, some of these desks are easily portable. Taking them to locations to record or even onstage is not half as daunting as humping a 40-channel analogue desk up a flight of stairs. Trust us, we've tried it.

And the down side?

It all sounds too good to be true, doesn't it? Well it's not all good news. Digital desks aren't as hands-on as their analogue forebears. An analogue desk has a button, fader or pot for every single function. There are no menus, no shared virtual controls and no real difference between operation on any two analogue desks. If you can work one, you can probably work them all... eventually; it's just a case of finding your way around. Digital desks rarely have that luxury. They all have different operation systems and often require a lot of 'playing with' before you get to grips with them. It's not unusual to discover functions you never knew existed on a digital desk, months after you got it.

Being able to hear something that needs adjusting, then doing it, is pretty much instantaneous on an analogue desk, but with digital it can take a few button presses before you get there. This might not bother you, but it's the main stumbling block for many brought up on analogue desks. Then there's maintenance. Digital desks don't require half as much, but when they go wrong, you really are shafted. It's often a case of back to the manufacturer. At least with analogue you can work on if a channel goes down or a fader dies.

Last (but certainly not least) there's the sound itself. Analogue desks are often warmer sounding due to their analogue circuitry. Digital desks, in theory, add no characteristics to the sound whatsoever. A common practice is to overdrive desk channels for subtle distortion, but with digital it just sounds nasty. This is one of the reasons for the popularity of valve outboard and channel strips at the moment. You can add that warmth to the signal either via a mic amp before the signal reaches the desk or via an inserted channel during the mix.

Digital domain

Love it or loathe it, digital mixing is here to stay. The advantages are just too numerous for digital desks not to catch on, big time. As it is, many home studios are based around HDRs with built-in digital mixers, so the newcomer to recording is more likely to understand the concepts of a digital mixer than an analogue.

Sean Vincent has seen this first hand: "I remember working with an engineer once who was freaking out because he had to use a 32-input analogue desk and he'd never used one before. He'd only ever used a Yamaha 02R at college. All those hands-on controls led him to believe it was going to be a nightmare. The strange thing is, we had to explain how to use it by comparing the bits of the analogue desk to screens on his digital one. Is it just me, or is that weird?"

Coming from an analogue background gives you a good insight into signal flow and routing, and we believe that's where you should start. How else are you going to appreciate the advantages of digital?

The big threat to digital mixers isn't analogue now anyway, it's software. Pro Tools users using something like ProControl or the Mackie HUI don't need a mixer at all. Even the latest version of Logic has 'live inputs' so you can feed your audio directly into the mixer within the software. Working this way, you only need to get more analogue inputs on your soundcard, and systems like the Digi 001 are making this cheaper all the time. The new Control 24 for Pro Tools from Focusrite is another take on the future of mixing, a control surface for the software as well as analogue inputs for your external gear. The best of both worlds or just a compromise? We'll have to wait and see.

Whatever mixer you choose, make sure you use your ears. Mixing is about music, not technology.

Sean Vincent and Alan Branch

The Soundcraft Guide to Mixing

Soundcraft

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SOUNDCRAFT
HARMAN INTERNATIONAL INDUSTRIES LTD.
CRANBORNE HOUSE,
CRANBORNE ROAD,
POTTERS BAR,
HERTS, EN6 3JN, UK.
TEL: +44 (0)1707 665000
FAX: +44 (0)1707 660742
EMAIL: info@soundcraft.com

SOUNDCRAFT USA
1449 DONELSON PIKE,
NASHVILLE TN 37217, USA
TEL: 1-615-360-0471
FAX: 1-615-360-0273
EMAIL: soundcraft-usa@harman.com

www.soundcraft.com

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THE SOUNDCRAFT GUIDE TO MIXING – CONTENTS

SECTION 1: STARTING OUT

SECTION 2: CONNECTING EQUIPMENT TO YOUR MIXER

SECTION 3: MIXING TECHNIQUES

SECTION 4: PA MIXING

SECTION 5: OTHER APPLICATIONS

SECTION 6: IN THE STUDIO

SECTION 7: WIRING UP & CONNECTORS

SECTION 8: GLOSSARY

STARTING OUT

A. What does a Mixer do?

No matter how sophisticated or expensive, all mixers carry out the same basic function - to blend and control the volume of a number of input signals, add effects and processing where required and route the resulting mix to the appropriate destination, which could be power amplifiers, the tracks of a recording device - or both. A mixer is the nerve centre of these sources, and therefore the most vital part of your audio system.

B. Guidelines in Choosing a Mixer

Audio mixers come in many different sizes and at all price levels, so it's little wonder that people are confused as to what type is actually needed for the job in hand. However there are several questions you can to ask yourself that will help you narrow your search to the most appropriate models.

- What am I going to be using the mixer for - i.e. multitrack recording, live PA work or both?
- What is my budget?
- How many sound sources do I have? As a guideline your mixer needs to have at least as many inputs as sound sources. If you are likely to be buying more equipment in the future you should budget for extra inputs.
- What particular mixer facilities **must I have** for my application? i.e. plenty of EQ, auxiliaries, or Direct Outs for recording.
- How portable does the mixer need to be?
- Will I be doing any location work where there won't be any mains power available?
- Have I read the Soundcraft Guide to Mixing from cover to cover?

Once you can answer these questions satisfactorily you should have a fairly accurate specification for the mixer you need.

C. The Controls - A Description

This is where we get into the nitty-gritty of what controls and inputs/outputs you'll find on a typical mixer. For this example, we've used a Spirit SX . **If you are already familiar with what the controls on a standard mixer do, then it's OK to skip to section 2.** If you find a term particularly difficult, further explanation can be found in the Glossary (Section 8).

MONO INPUTS

A Mic In

Use this "XLR" input to connect your microphones or DI boxes.

For Mic Input Wiring Explanations see section 7.

B Line In

Use this connector for plugging in "Line Level" instruments such as keyboards, samplers or drum machines. It can also be used to accept the returns from multitrack tape machines and other recording media. The Line Input is not designed for microphones and although it may be used, will not provide optimum performance with them.

For Line Input wiring explanations see section 7.

C Insert Point

This is used to connect external signal processors such as compressors or limiters within the input module. The Insert Point allows external devices to be placed within the Input Path - see Fig. 1.1.

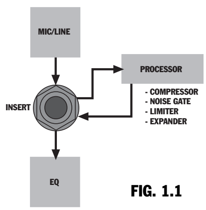
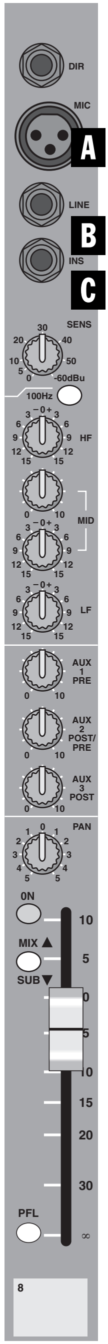


FIG. 1.1

See Section 2 and 3 for more detail on how to use processors, and Section 7 for information on wiring.



D Direct Out

This allows you to send audio direct from your channel out to a multitrack tape recorder, or to an effects unit when the channel requires its own special effect.

See sections 2 and 6 for more details on connections and studio techniques.

E Gain Control (Input Sensitivity)

Sets how much of the signal from the mic or line inputs is fed to the channel.

F HPF (High Pass Filter)

As the name suggests this switch cuts out the very lowest frequencies of a sound whilst allowing the higher frequencies to "Pass Through". It's particularly useful in live situations to reduce stage rumble or microphone 'popping', which can produce a muddy mix, or to 'clean-up' male vocals and filter out low frequency hum. Some manufacturers may also use the term "low - cut" filter to describe the HPF. See Fig. 1.2.

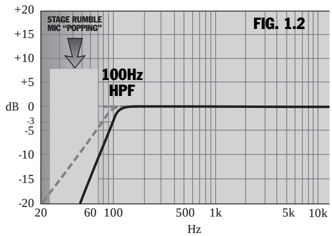


FIG. 1.2

G EQ Section

Usually the most closely scrutinised part of any mixer, the equaliser section allows you to change the tone of the sound on each input. An EQ is normally split into "bands", which control a range of frequencies, in a similar fashion to the treble and bass tone controls on your Hi-Fi. Indeed a simple "2 band" EQ is little more than an input treble and bass control. The more bands an EQ has the more sophisticated it is. SX has a 3 band EQ, with a separate control for the middle audio frequencies. This control is also "swept" which provides even more sophistication. Simply described, a sweep EQ allows you to choose the exact frequency to cut and boost, rather than having it chosen for you, as on normal "fixed" controls.

We will talk in more detail about EQ in section 3.

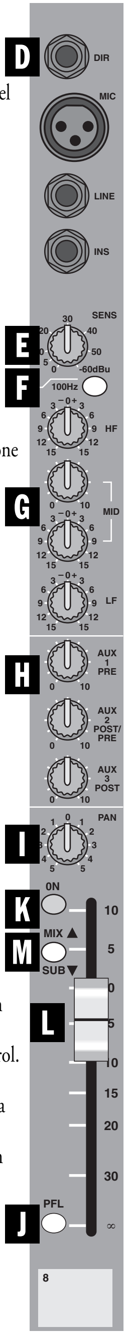


FIG. 1.3

H Auxiliary Section

Typically, these controls have two functions: First, to control the levels of *effects* such as reverb from external effects units that have been added to the input signal, and second to create separate musician's *"foldback"* mixes in the studio or on stage.

How to use auxiliaries, connecting them to external equipment and other applications are described in section 3.

I Pan (Panoramic Control)

This determines the position of the signal within the stereo mix image or may be used to route (send) the signal to particular GROUP outputs as selected by the ROUTING SWITCHES (see below).

J Solo (PFL and Solo in Place)

The PFL solo switch allows you to monitor an input signal independently of any other instruments that have been connected, which is useful for troubleshooting, or setting an instrument's Input Preamp Gain and EQ setting.

Pre-Fade Listen (PFL) is a type of solo that allows you to monitor your sound BEFORE THE FADER. In other words when you move the input fader in PFL mode the level will not change, nor will you hear any effects. Because effects and volume are not a distraction, PFL solo is very useful for setting proper input preamp levels.

Some Soundcraft mixers use SOLO IN PLACE, which allows you to monitor signals after the fader in their true stereo image, and with any effects that have been added. This type of Solo is less good for level setting, but more useful in mixdown situations for auditioning sounds.

See section 3 - Setting Gain, for more information on using PFL.

K Mute/Channel On-Off Switch

This turns the channel on or off and is useful for isolating the channel when not in use or pre-setting channel levels which may not be needed until later, ie: theatre scene-setting or support acts/performers.

L Fader

This determines the level of the input signal within the mix and provides a visible indication of channel level.

M Routing

By selecting the routing switches the input signal is sent to a choice of the mixer's outputs - typically the main mix outs or the group outputs. The switches are used in conjunction with the PAN control to route the signal proportionately to the left or the right side of the mix or to odd/even groups/subs if PAN is turned fully left or right.

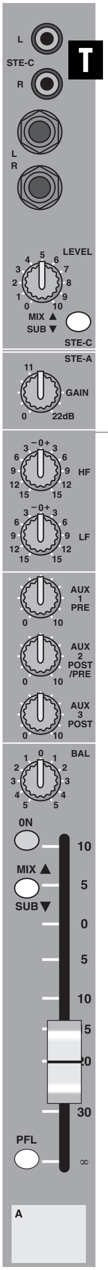


FIG. 1.4

STEREO INPUTS

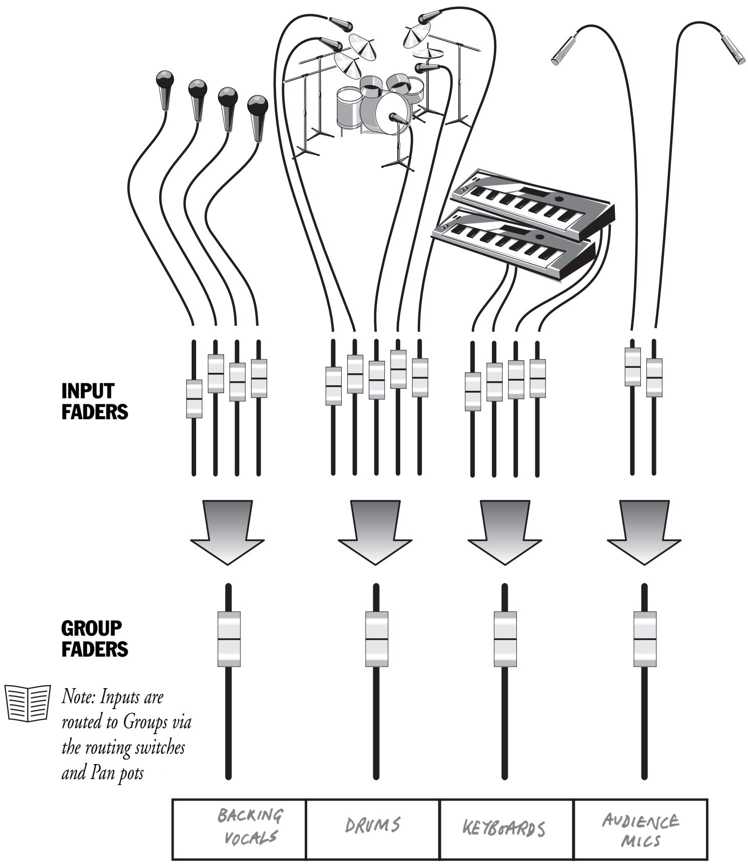
Guitar amps and mic'd sound sources only provide you with mono signals. However keyboards, samplers, drum machines and other electronic media often provide true stereo outputs with separate left and right signals. Stereo Inputs on mixers simply allow you to connect both of these signals individually and control them from a single fader. Stereo inputs tend to incorporate fewer facilities than mono inputs as most keyboards are already equipped with plenty of internal effects and tone control options.

NB: Soundcraft jack stereo inputs default to Mono when the left input is used. RCA phono connectors do NOT have this option.

SUBGROUPS

These allow the logical assignment of groups of instruments or vocalists so that they may be controlled by just one pair of faders, or even a single fader, once individual instruments' relative levels have been balanced. They also act as additional outputs with separate volume/level controls – ideal for speaker fills or recording a number of instruments to one tape track.

FIG. 1.5



THE MASTER SECTION

N Mix Outputs
Mix outputs provide left and right level control of the final stereo mix. Many consoles feature mix insert points too, allowing the connection of signal processors across the whole mix.

O Monitor "Engineer's" / Control Room Outputs
These let you listen to any solo, mix, submix from a group, or the 2 Track tape return via an external amplifier and speakers, or the headphone socket.

P 2 Track Tape Returns
Allow you to connect the outputs of your cassette or DAT player and listen back to your completed masterwork. They may also be used for playing pre-show music at a gig using 2-Track to Mix switch (not shown in illustration).

Q Aux Masters
These govern the overall output levels from the auxiliary outputs and therefore the amount of signal going to an effects unit or a musician's foldback mix.

R AFL
Allows monitoring of the actual signal leaving the Aux Masters.

S Meters
Normally they show mix output levels. When any Solo button is pressed, the meters automatically switch to show the solo level. They provide visual indication of what's going on in your mixer.

T Stereo Returns (see *Stereo Inputs* earlier in this section)
These allow signals from external equipment, such as effects units, to be returned to the mixer and routed to the stereo Mix or Groups, without using up valuable input channels.

U +48v or Phantom Power
Some microphones, known as condenser mics, require battery power to operate. Alternatively the power may be provided by the console. This is known as 'phantom power' and runs at 48vDC. Simply press "Phantom Power" and any condenser mics connected will operate without the need for batteries.

Caution: DO NOT ACTIVATE A GLOBAL PHANTOM POWER SWITCH IF AN UNBALANCED SIGNAL SOURCES IS CONNECTED TO ANY MIC INPUT. Because of the voltage present on pins 2 and 3 of the XLR connector, you will damage your microphone/signal source.

Always refer to your Mixer's User Guide.

More Information on Condenser Mics can be found in Section 3 - Mixing Techniques.

Further detail on mic wiring may be found in Section 7 - wiring.

V Headphones
Allow you to listen to your mix without annoying your neighbours or being distracted by ambient sounds.

That's it, the basic features of your average mixing console. If you found it a little heavy going, don't despair: it does get easier!

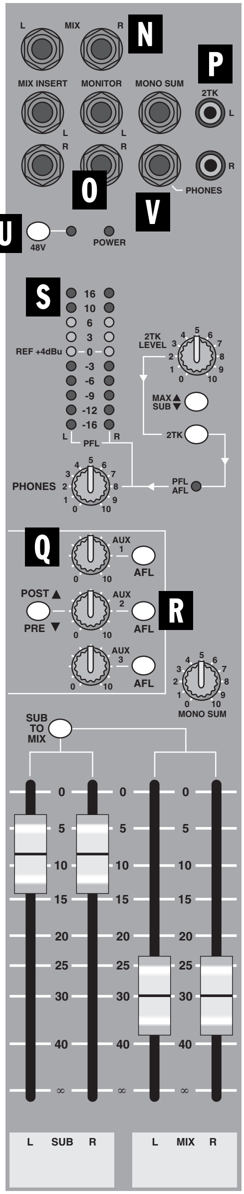


FIG. 1.5

D. Signal Flow

Now the typical mixer features have been explained in detail it is important to understand how they form together. The route which a signal source takes through a mixer is normally shown using one of two devices: a **block diagram** or a **signal flow diagram**.

Both diagrams provide a 'visual' description of the key elements of the mixing console. They allow you to identify which components are used in the audio path and help the engineer to "troubleshoot" when signal sources don't appear to be doing what they should! In simple terms, they are electronic maps.

An example of a signal flow diagram is shown here. This is the most basic representation of console layout, showing a how a single sound source may pass through an input strip to the various other parts of the mixer.

Block diagrams are slightly more complex, showing more detail, electronic information, including the location of resistors and capacitors, and the structure of the entire console including bussing: **an example is shown on page 37**. Block diagrams also use a number of symbols to represent electronic elements. A few minutes spent understanding them some time during your journey through this booklet will most definitely pay-off in future mixing projects.

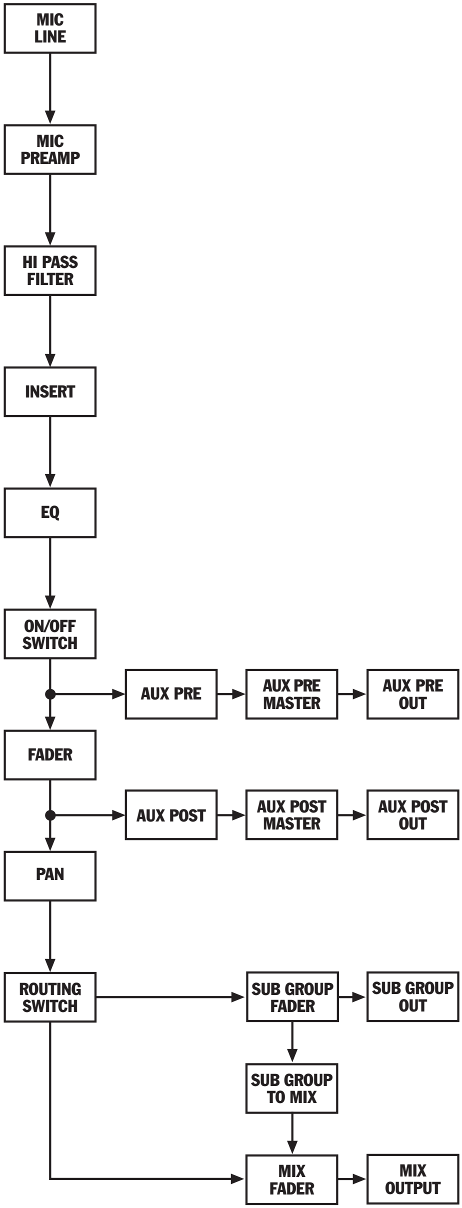


FIG. 1.6

A Typical Signal Flow Path

CONNECTING EQUIPMENT TO YOUR MIXER

As we explained in the last section, it is the job of the mixer to accept the various signal sources, set the levels and route those signals to the correct destination.

We'll now take a quick look at where to connect the 'peripheral' equipment that you will be using with your mixer. If you have already created your own set-ups successfully in the past, you should only need to skim this part.

A. Input Devices

Microphones

All microphones should be connected via each input's XLR connectors. Do *not* use line inputs.

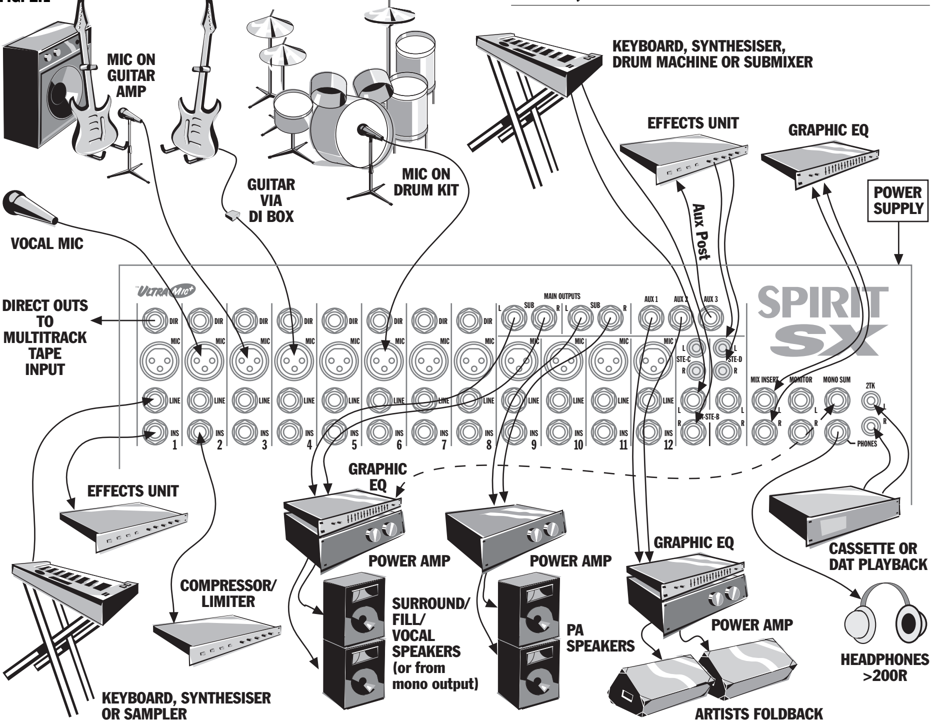
For more information on miking up individual instruments, refer to sections 4 and 6 - PA Mixing and In the Studio.

Direct Injection Box (DI Box)

- A DI Box allows you to connect a guitar or bass directly to the mixer's input, rather than miking up the instrument's amp/speaker. This technique is often preferred by musicians who require a "clean" sound. The best DI boxes are ACTIVE and require Phantom Power like condenser microphones. They should be connected to XLR mic inputs.

NB: Although electric guitars and basses may be connected to a mixer's line inputs without danger, the results will be far from ideal, because the IMPEDANCE of these instruments will not match up with typical line levels. Direct connection usually leads to a weak sound.

FIG. 2.1



Electronic Line Output Devices

- Keyboards, Drum Machines, CD Players, DAT Machines, Wireless Mic Receivers, all provide line level outputs, and should all be connected straight into the Mixer's Line Inputs. If some of your instruments are STEREO connect their left and right outputs to a spare stereo input. Alternatively connect to an adjacent pair of mono inputs and Pan the inputs hard left and right to create a stereo image.

B. Equipment Requiring Both Inputs and Outputs

External effects units

Connect the input of your effects unit marked "mono" to A POST FADER AUXILIARY OUTPUT. If you are uncertain, Post fader auxiliaries are coloured blue on Soundcraft mixers with the relevant channel aux pots usually marked "post". The left and right outputs from the effects unit should be connected to a pair of stereo returns, or stereo inputs if stereo returns are not available. If intensive EQ is required, use a pair of Mono Inputs. Remember, the effects signal is no different from any other audio signal – it still requires an input to the mixer.

See Section 3 Mixing Techniques or a detailed explanation of post fader auxiliaries.

NB: YOU DO NOT HAVE TO CONNECT UP BOTH THE LEFT AND RIGHT INPUTS OF YOUR EFFECTS UNIT TO SEPARATE AUXs. Most units only require "pseudo-stereo" operation and will mimic a stereo reverb or effect inside before providing a stereo output to the mixer's returns.

Signal Processors

Connect signal processors, such as compressors to the insert jack using a special insert 'Y' cable. This allows the signal to be sent and returned to the mixer using only one connector.

Refer to section 7 for wiring information.

It is also possible to connect the processor to the console without using the insert jacks by connecting an instrument direct to the processor first. However, the advantage of using processors in the mix/group or channel inserts is that any level changes made by the processor can be monitored by the mixers meters.

NB: A signal processor can be used in a channel to control one audio source, across a group to control a number of audio sources or across the entire mix.

Tape machines

Multitrack machines are used for initial track-laying in either studio or live recording situations.
For more sophisticated work, a stand-alone machine offers better sound quality and greater versatility than a cassette multitracker. The new generation of digital multitracks are also very attractive, but analogue, open-reel multitracks are also capable of professional sounding results. Aim for a minimum of eight tracks if your budget will allow.

Mastering Machines

Your final mix should be recorded on the best quality machine that you can afford. A recording is only as good as the weakest link in the chain, and a good cassette machine is fine for demos, but for more serious work, consider a DAT machine or perhaps a second hand, open-reel 2-track.

C. Output Devices

Amps and Speakers (Monitor and FOH)

Studio Monitoring

A high-powered hi-fi amp of around 50 watts per channel is fine for home recording, but to ensure adequate head-room you should consider a well-specified rack mount amp. Similarly, a pair of accurate hi-fi speakers will do the job, but for more serious work we would recommend purpose-designed nearfield monitors. Always remember that no matter how good the recording or performance, a poor monitoring set-up will not allow you to make qualitative judgements about the mix.

Headphones

When choosing headphones for monitoring, you'll obviously want a pair that give the best sound reproduction for the price. But, bear in mind that in order for you to fully concentrate on the mix, the headphones should exclude outside noise - therefore open-back designs will be of little use.

Furthermore, you could be wearing the headphones for several hours at a stretch so comfort is essential.

NB: Make Sure that the IMPEDANCE of your headphones matches the specification of your mixer.

PA Work

PA work requires high-powered, rugged, and honestly specified amps and FOH (Front of House) speakers. The power rating of the system will depend on the size of venues you will be playing.
See PA Mixing, Section 4, for more information.

MIXING TECHNIQUES

A. Choosing the Right Microphone

Microphone Types

The choice of microphone depends on the application that the microphone will be used for and individual preference. However, broadly speaking microphones fall into two main types:

Dynamic Microphone -

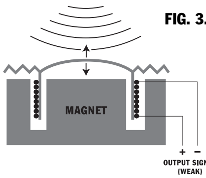


FIG. 3.1

• A robust design which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.
Dynamic mics are relatively inexpensive, rugged and require no electrical power to operate. They are ideal for all-round high sound pressure levels (SPL) and tend to be used for live applications. However, they are not as sensitive to high frequencies as condenser types.

Condenser Microphone -

• A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. They need power to operate - the most common source being +48v DC PHANTOM POWER.
Condenser mics are very sensitive to distant sounds and high frequencies. Because of this sensitivity they are often used in studio recording situations.

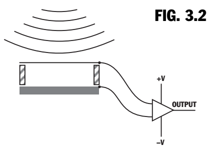


FIG. 3.2

N.B. +48v Phantom power is used to charge the diaphragm and plate. It also supplies a small amplifier which boosts the small voltages generated by diaphragm movements.

Microphone Pick-up Patterns

A pick-up (Polar) pattern refers to the area(s) from which a microphone "picks up" its sound. It is important to choose the right pattern for your application, or you may pick up sounds from areas you don't want or lose sound information you need.

Omni Pattern

The most basic type of microphone pattern.


FIG. 3.3

• A 360° polar response which picks up sound equally in all directions.
This pattern is ideal for picking up groups of vocals, audiences, ambient sounds but is most susceptible to feedback.

Cardioid Pattern

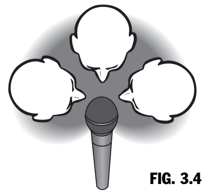


FIG. 3.4

• The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front.
Used for most basic recording or in any situation where sound has to be picked up from mainly one direction. Dynamic cardioid mics are mostly used for live applications because they help reduce unwanted spill from other instruments, thus reducing the risk of feedback.

Hyper-cardioid


FIG. 3.5

• Similar to a cardioid pattern but with greater directionality.
Used for live vocal microphones because it provides the greatest protection from unwanted spill and feedback.

Figure of Eight

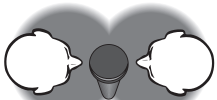


FIG. 3.6

• Sound is picked up from the front and back but not from the sides.
This pattern is used mainly in studios for picking up two 'harmony' vocalists, or solo vocalists who require some room ambience.

B. Setting Up a Basic Mix

Setting the Gain

Input gain is designed to take an audio signal, and adjust it to the level which the mixer understands.

All audio circuits, mixers included, produce a low level of electronic noise or hiss, and while this can be made very low by careful design, it can never be completely eliminated. It is also true that any audio circuit can be driven into distortion if the input is too high in level; hence care has to be taken when setting the input level so as to preserve the best possible sound quality. Ideally the input signal should be as high in level as possible while still leaving a margin of safety to prevent distortion on loud sections. This will ensure that the signal is large enough to render the background noise insignificant, whilst keeping the signal clean. The remaining safety margin is known as *Headroom*.

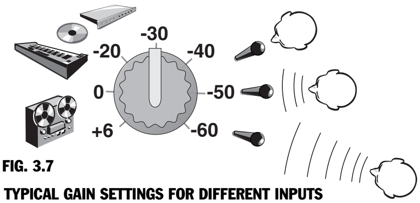


FIG. 3.7

TYPICAL GAIN SETTINGS FOR DIFFERENT INPUTS

To set the gain on the mixer;

- Press the PFL/Solo switch on the relevant input.
- Adjust gain/input sensitivity until meters read within the yellow ('3' to '6' on meter scale). This allows for the extra 10dB of gain that is available on Soundcraft input faders.
- Release PFL/Solo.
- Repeat for all other inputs.

NB: EQ affects gains settings. If you adjust the EQ you will need to re-check your gain level using the above method.

Once you have optimised the gain your mixer will give the best possible signal quality with the minimum of noise and distortion.

Balancing Fader Levels

Faders allow you to make fine adjustments to your sounds and act as a visual indication of the overall mix levels.

It is important to keep your input faders around the '0' mark for greater control. This is because fader scales are typically logarithmic and not linear, so if your fader position is near the bottom of its travel then even a small movement will lead to huge leaps in level. Similarly try not to have your fader at the top of its travel because this will leave you no room to further boost the signal.

See diagram below.

Balancing Output Levels

Master Outputs

Set your master outputs to '0' on the scale. There are two reasons for this:

1 You have the maximum fader travel for fading out your mix.
2 If your faders are set below '0' you will not be getting the full benefit from the meters because you will only be using the first few LEDs on the meter scale.

NB: Your mixer is not an amplifier. So the master output faders should be set to maximum ('0' on scale). If extra output is required, then turn up your amplifier.

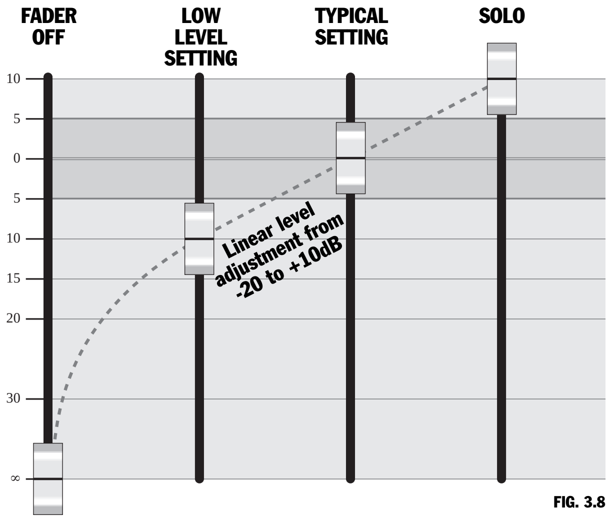


FIG. 3.8

C Using the Mixer's EQ

Equalisation is useful for making both corrective and creative changes to a sound, but it needs to be used with care. Corrective applications include making tonal changes to compensate for imperfect room acoustics, budget microphones or inaccurate loudspeaker systems. While every effort should be made to get the sound right at source, this is less easily achieved live than in the more controlled conditions of the recording studio. Indeed, the use of equalisation is often the only way to reach a workable compromise in live situations.

Creative applications, on the other hand, are equally as valid in the recording studio as they are live, and an equaliser with a swept midrange control is infinitely more versatile than one that has simple high and low controls. The only rule of creative equalisation is - 'If it sounds good, it is good!'

Fixed EQ

Most people will be familiar with the operation of high and low frequency controls; they work in a similar manner to the tone controls on a domestic stereo system.

In the centre position the controls have no effect, but rotate them clockwise and they will provide boost, or rotate them anticlockwise and they provide cut. Despite their apparent simplicity, however, high and low controls should be used with caution as overuse can make things worse. Adding a small amount of high or low boost should be enough to add a touch of brightness or warmth to a sound, but a quarter of a turn should be sufficient, especially where the low control is concerned.

The drawback with fixed controls often lies in the fact that you may want to boost just a particular sound such as the punch of a bass drum or the ring of a cymbal, whereas a fixed control influences a relatively large section of the audio spectrum. Apply too much bass boost and you could find the bass guitar, bass drum and any other bass sounds take on a flabby, uncontrolled characteristic which makes the mix sound muddy and badly defined. This is because sounds occupying the lower mid part of the spectrum are also affected. Similarly, use too much top boost and the sound becomes edgy with any noise or tape hiss being emphasised quite considerably.

In a PA situation, excessive EQ boost in any part of the audio spectrum will increase the risk of acoustic feedback via the vocal microphones.

THE FREQUENCY RANGE OF DIFFERENT INSTRUMENTS AND WHICH EQ BANDS AFFECT THEM

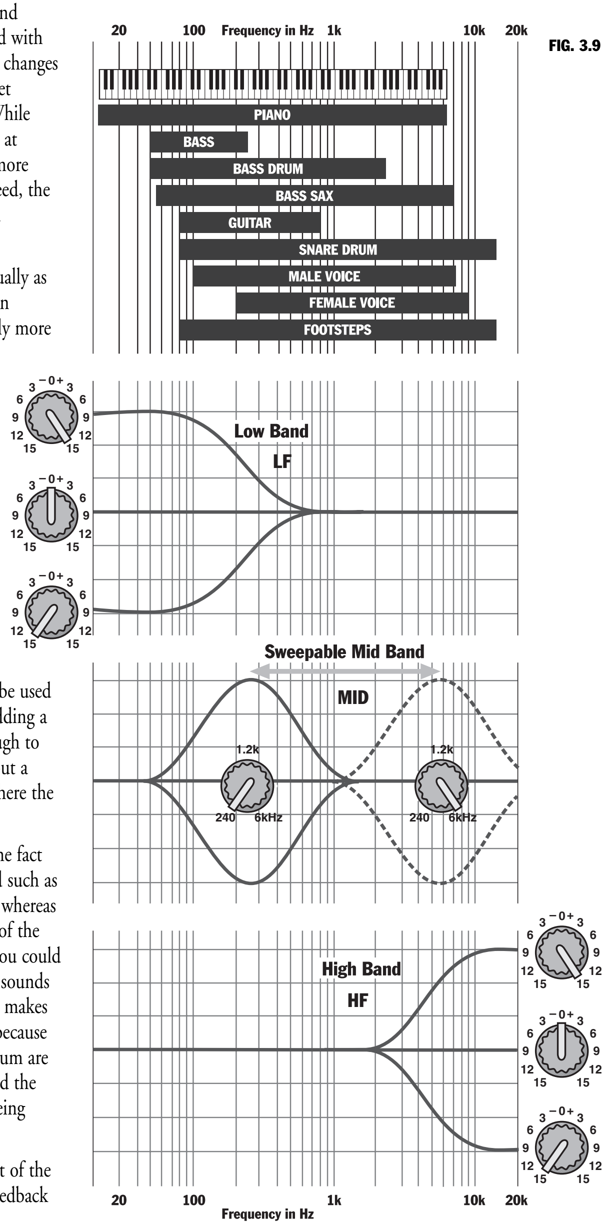


FIG. 3.9

Bearing the above points in mind, the best approach is to use small amounts of boost, especially when working live. EQ cut, on the other hand, causes far fewer problems, and *rather than boost a particular sound it is frequently more rewarding to apply cut in whichever part of the audio spectrum that appears to be overpowering*. In this application, the sweep mid control is also very effective.

Using a sweep-mid equaliser

Like the high and low controls, the sweep mid can provide either cut or boost, but its strength comes from the fact that it can be 'tuned' into the specific part of the audio spectrum that needs treatment. Like the high and low controls, it is more forgiving if used to cut rather than to boost. However, when first tuning in the mid control, it helps to set it to full boost, so that when the frequency control is adjusted, the effect is most apparent. This is true even if the final EQ setting requires cut rather than boost.

Procedure

Below is a simple way of eliminating unwanted sounds:

> **!** *Caution: when adjusting EQ, there is a danger of feedback which can cause damage to your speakers. You may need to reduce your levels to compensate.*

- Increase sweep-EQ gain.
- Sweep the frequency pot until the aspect of the sound you wish to modify becomes as pronounced as possible. This should only take a few seconds.
- The cut/boost control is now changed from its full boost position to cut. The exact amount of cut required can be decided by listening to the sound while making adjustments.
- Even a small amount of cut at the right frequency will clean up the sound to a surprising degree.

Other sounds may benefit from a little boost, one example being the electric guitar which often needs a little extra bite to help it cut through the mix. Again, turn to full boost and use the frequency control to pick out the area where the sound needs help. Then it's a simple matter of turning the boost down to a more modest level and assessing the results by ear.

D. Using Effects Units

The Different Types

The problem with mixing 'dry' (using no effects) within a live or recording environment is that the results can often sound boring and lacking in colour. This is especially the case as most of us are used to listening to highly polished CDs at home. These productions are actually achieved by using effects which electronically produce certain atmospheres. The different types of effects that can be used are explained below;

Reverb

Reverberation is the most commonly used studio effect, and also the most necessary. Western music is invariably performed indoors where a degree of room reverberation is part of the sound. Conversely, most pop music is recorded in a relatively small, dry-sounding studio, so artificial reverberation has to be added to create a sense of space and reality. Reverberation is created naturally when a sound is reflected and re-reflected from the surfaces within a room, hall or other large structure. See fig. 10.

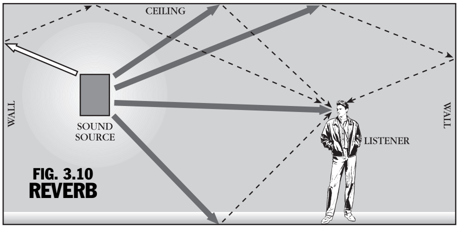


FIG. 3.10
REVERB

Delay

Often used to make a sound 'thicker' by taking the original sound, delaying it, then mixing it back with the original sound. This short delay added to the original sound has the effect of doubling the signal.

Echo

A popular effect that was used extensively on guitars and vocals in the 60s and 70s. It is not used on vocals so much nowadays, but quite effective on guitars and keyboards. A neat trick is to set the echo delay time so that the repeats coincide with the tempo of the song.

Chorus & Flanging

Both chorus and flangers are based on a short delay, combined with pitch modulation to create the effect of two or more instruments playing the same part. Flanging also employs feedback and is a much stronger effect. Both these

treatments work well on synth pad sounds such as strings and are best used in stereo where they create a sense of movement as well as width.

Pitch Shifters

These change the pitch of the original signal, usually by up to one octave in either direction and sometimes by two. Small pitch shifts are useful for creating de-tuning or doubling effects. Which can make a single voice or instrument sound like two or three, while larger shifts can be used to create octaves or parallel harmonies.

NB: For useful effects settings with different instruments refer to Section 6 'In the Studio'.

Setting up an effects loop

- Set effect unit to full 'wet' signal
- Connect your effect units as per Section 2, Input Devices.
- On the relevant input channel, set the post fade aux to maximum
- Select AFL on your aux master
- Set aux master level so that the meters read '0'
- Adjust input level on effects unit until 'effects meters reads '0' (nominal)

NB: You can now use the mixer AFL meters to monitor effects unit levels as both meters have been calibrated.

- Release aux master AFL and select effects return PFL

NB: If you are using a simple stereo input with no PFL, adjust input gain for required effect.

- Adjust effects return input gain until meters read around '0'.
- De-select PFL and adjust effects return fader level for required effect level.

NB: The original 'dry' signal is mixed with the effects 'wet' signal.

Pre- and Post-fade Auxiliaries

Pre-Fade

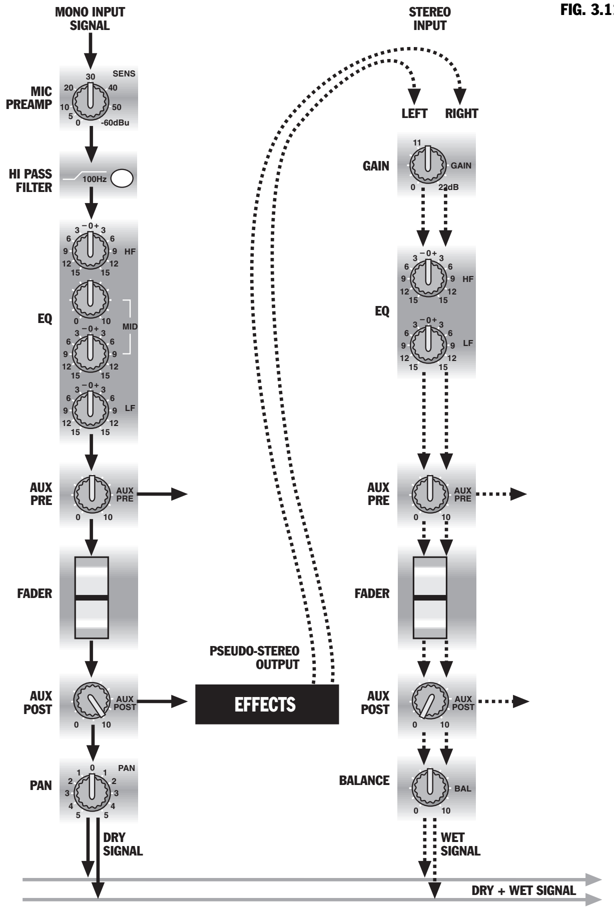
Pre-fade auxiliaries are independent of the fader so that the amount of effect will not change with new fader levels. This means you will still hear the effect even when the fader is at the bottom of its travel.

Post-Fade

It is important to use post fade auxiliary sends for effects units. This is because post fade auxiliaries 'follow' the input fader so that when input level changes the amount of effect remains proportional to the new input level.

NB: Effects Return Aux Post Control must be set to minimum or feedback will occur

FIG. 3.11



E. Using Signal Processors

The Difference between Signal Processors and Effects

Unlike effects, which are creative in nature, signal processors are used to control and manipulate sounds to achieve the best audio quality performances and recordings.

Effects and signal processors should never be confused. Whereas effects are "mixed" with an input to provide a combined sound, signal processors alter an input, group or mix signal completely. The signal is actually taken out of the mixer entirely, "processed" and returned in its altered state, in series with the original audio signal.

For this reason signal processors should be connected using Insert Points and not the Auxiliary Send and Return Loop (effects loop).

NB: Effects can be connected to inserts if necessary, but then the proportion of the effect in the signal is governed solely by the effects unit mix control.

The Different Types of Signal Processors

Broadly speaking, there are 5 different types of signal processor in common use:

Graphic Equalisers

Graphic Equalisers work by splitting the sound spectrum into narrow, adjacent frequency bands and giving each band its own cut/boost slider. The term Graphic comes about because the position or 'curve' of the sliders gives a graphic representation of the way in which the settings affect the audio frequency range.

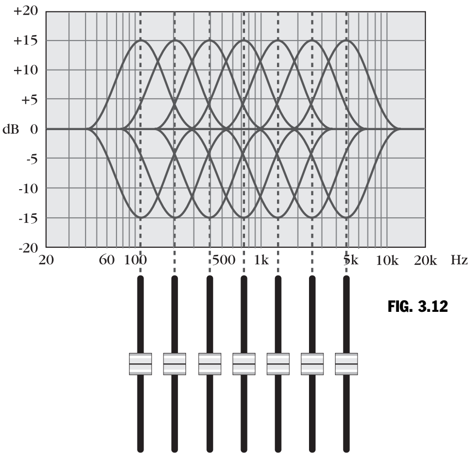


FIG. 3.12

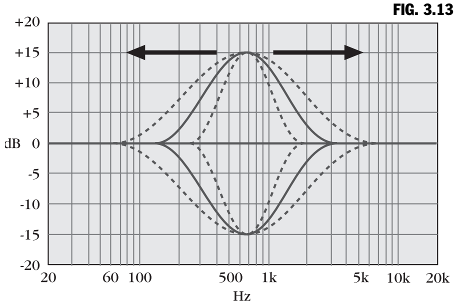
Graphic Equalisers are most often used to process the mix at live venues by notching out troublesome frequencies that may be causing feedback. They may also be used to enhance a mix at a poor sounding venue. In recording they are used to create "flat" listening environments.

For more detail on venue acoustics go to section 4 - PA Mixing.

Parametric Equalisers

These are similar to the EQ found on an input channel but may include more bands and additional bandwidth (Q) controls which define how many frequencies in the band are affected.

FIG. 3.13



They are most often used to provide additional creative control over an input signal when a mixer's EQ is not sufficient.

Gates

A gate is designed to shut down the audio signal path when the input signal falls below a threshold set by the user. It may be used to clean-up any signal that has pauses in it. For example gates are widely used to prevent 'spill' between adjacent mics on a multi-mic'd drum kit where, say, a tom-tom mic may pick up the snare drum.

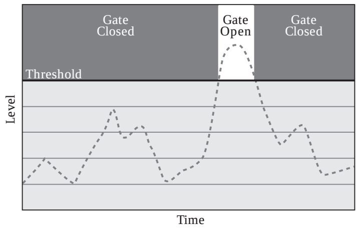


FIG. 3.14

Expanders

Expanders accomplish much the same task as gates, though they are more like compressors in reverse. Compressors affect the gain of signals exceeding the threshold, while expanders act on signals falling below the threshold. A gate will close completely when the signal falls below its

threshold, but an expander works like an automatic mixing engineer who pulls down the signal when the signal falls below the threshold; the more it falls below the threshold - the more he pulls down the fader.

Expanders are most often use in Studio recording to provide the best mix signal to noise ratio when producing final masters.

Compressor/Limiters

A compressor reduces the difference between the loudest and quietest parts of a performance. It works on a threshold system where signals exceeding the threshold are processed and those falling below it pass through unchanged. When a signal exceeds the threshold the compressor automatically reduces the gain. How much gain reduction is applied depends on the 'compression ratio' which on most compressors is variable: the higher the ratio, the stronger the compression. Very high ratios cause the compressor to act as a limiter where the input signal is prevented from ever exceeding the threshold.

Compressors are the most commonly used processor and are particularly popular for maintaining constant vocal and bass guitar levels live and in the studio. This is because, out of all instruments, singers tend to vary their levels the most. Compressors help to achieve the much sought-after tight, "punchy" sound.

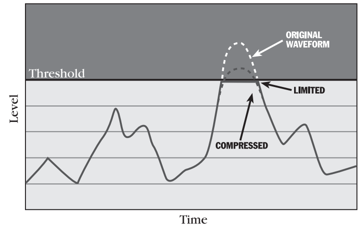


FIG. 3.15

Setting up a Signal Processor

- Connect your processor to the relevant mixer insert jack (mono, group or mix insert), using a insert 'Y' lead.

Refer to section 7 for wiring information

- Set your processor to unity gain (x1), i.e. no additional gain.
- Make your adjustments on your signal processor
- Beware that your processor settings may alter your mixer input output levels. Re-adjust levels to '0' on meters, if necessary.

NB: Remember a signal processor can be used in a channel to control one audio source, in a group to control a number of audio sources, or to control the entire mix.

F. Creating a Foldback/ Monitor Mix

Performers usually require their own mix independent from the main/engineer's mix. This is because to achieve the optimum performance they need to hear themselves above other voices or instruments. This performer's mix is known as a foldback/monitor mix.

The procedure is as follows;

- Set the **pre-fade** aux to maximum on the relevant performers input channel.
- Select AFL on your aux master.
- Set aux master level so meters read '0'.
- Create a foldback mix for the performer by setting the pre-fade aux levels on the other performer's input channels.
- Release aux master AFL.

NB: It is typical that the performers' own vocals/instruments will be two thirds louder than any other sources in their own monitor mix.

Each performer may require a separate monitor mix/auxiliary output.

NOTE: Pre-fade rather than post-fade auxiliaries must be used. This is because they are independent of the input faders. If post-fade auxiliaries are used, then foldback mix levels will alter with every input fader change made by the FOH engineer. This will annoy the band and may lead to feedback which can damage speakers and headphones.

Now that you know how to connect and set up different elements of your system let's look at some real-world examples of systems in use.

PA MIXING

A. A Typical Live Performance

Introduction

There are so many different types of 'live' scenarios that it would be almost impossible for us to describe each one in a book of these modest proportions. Instead, our 'typical live gig' is represented by a small band, whose set-up is shown in the "Mixing Live" diagram.

Microphones

Most of the microphones used in live applications are dynamic cardioids because they are tough, produce an intelligible sound and their directional response helps prevent spill or feedback. Dynamic microphones can handle anything from drums to vocals. However, condenser types, with their greater sensitivity to high frequencies are invariably used for jobs such as overhead pick-up on a drum kit or mic'ing acoustic instruments.

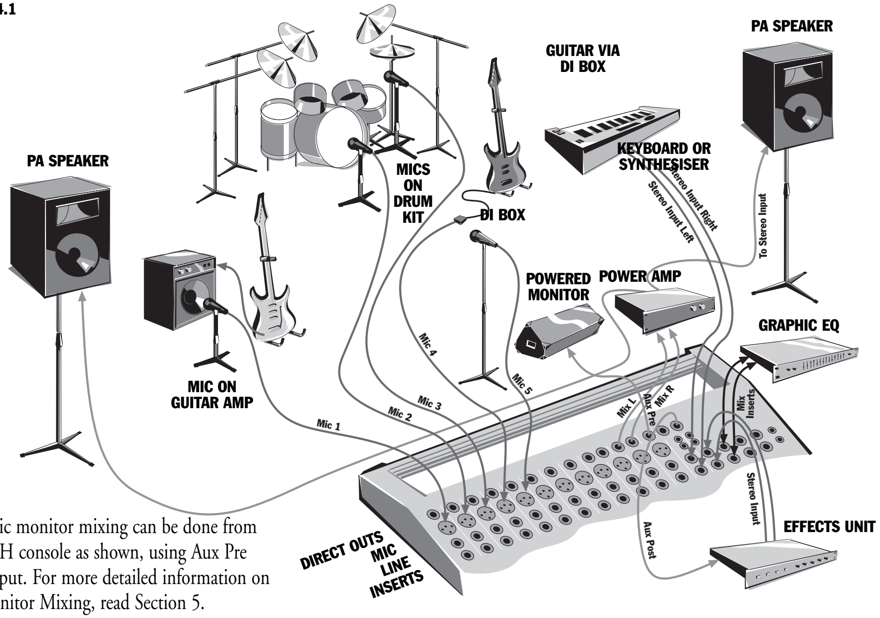
Cables and Connections

Interference and hum can be avoided! A few minutes spent checking cable runs and connectors pays dividends.

- A balanced audio connection provides low noise operation by cancelling out any interference in a signal. It does this by using a 2-conductor mic cable surrounded by a shield. Any interference picked up will be of the same polarity on the two conductors and is therefore rejected by the mic input's Differential Amplifier.
- Don't skimp on interconnecting cables - always buy the best that you can afford. Make sure that all connections are sound and keep cable runs as short as is practicable.
- A multicore cable and stage box will keep trailing cables to a minimum and presents a tidy and practical approach.
- If your mixer has a separate power supply unit, keep it well away from the console.
- Where signal and mains cables must cross, make sure they're at 90° to each other. This will help reduce the risk of hum and noise.
- If the venue has a three-phase supply, don't share the same phase as lighting controllers.

MIXING LIVE

FIG. 4.1



- Basic monitor mixing can be done from FOH console as shown, using Aux Pre output. For more detailed information on Monitor Mixing, read Section 5.

- It is dangerous to lift the mains earth when trying to eliminate hum. You can isolate hum by lifting the appropriate audio signal shield.
- When using wireless mics, set the receiver on stage and run back to the console at balanced mic level. This will help avoid interference from digital sources and lighting controllers.
- Keep unbalanced 'insert' leads away from mains and keep them short - no longer than about 2 metres.

Connecting External Effects and Processors

We talked about Effects and Processors in Sections 2 and 3, so you're now aware of their functions and applications. Effects units are best connected via the console's Auxiliary Send and Return Loop (sometimes known as the Effects Send and Return Loop) or the Insert Point. When used in the Aux Send system, the dry signal level should be turned off on the effects unit, but when used via Insert Points (for guidance on how to wire a jack for use with Insert Points, see Section 6), the dry/effects balance must be set on the effects unit itself. Processors treat the whole of the incoming signal and therefore may only be used via console Insert Points or directly 'in-line' with a signal: they cannot be used in the Aux Send/Return loop system.

Setting Up

- Position the mixing console so that you can hear the on-stage performance as the audience will hear it. Ensure that you have a clear view of the performers.
- After setting up, switch the power amps on last to prevent any thumps occurring when effects or instruments are powered up. Ensure the console's master gain is down before you switch on the amplifiers.
- Don't set up the vocal mic directly in front of the drum kit or a guitar stack.
- Make sure the speakers aren't obstructed by the audience and that the majority of the sound is being directed towards the audience, not towards the rear or side walls.
- Set up the vocal levels first - it's no use getting a great drum sound if the vocals feed back before they can even be heard.
- Keep the vocals panned towards the centre of the mix. Not only will this sound more natural, but it will allow the greatest vocal level before feedback or distortion occurs.
- Be sparing on the use of artificial reverb. Most venues are too reverberant anyway, and excessive reverb will ruin the intelligibility of the vocal performance.
- Do not use reverb on low frequency sound sources such as bass, kick drums and tom toms.
- Keep backline amp levels down: let the mic and mixer do the work!
- Always leave a little gain in hand so you can wind up the level slightly as the show progresses.
- Putting high levels of bass guitar or kick drum through a small PA can overload the system and distort vocal quality. Try rolling off some of the low bass, you'll get a higher subjective sound level without overload.

Ringing Out: Nulling Room Acoustics

! *Caution: Ringing out can cause howl around which can damage speakers, so use care when adjusting levels.*

As experienced engineers will tell you, there's no such thing as a perfect venue. To help tailor the sound to the room acoustics, insert a Graphic Equalizer into the console's mix insert jacks which are effectively between the mixer and the power amp.

'Ringing Out' the system prior to the sound check will help reduce troublesome feedback. To Ring Out, follow this procedure:

1 Set all graphic EQ controls to centre (0).

2 Turn up amp volume until feedback is just beginning to 'ring'.

3 Turn back the amp volume slightly to prevent accidental feedback.

4 Starting from the left, adjust the first graphic EQ frequency gain control to 'max': if the system doesn't feedback, then this is not a problem frequency. Return this gain control to centre position.
If the system feeds back, reduce the EQ gain by the same amount you boosted to get feedback.

5 Repeat this procedure for all graphic EQ frequencies.

Setting the Mix

- Turn down the amplifier gain before the system is first switched on. This will avoid unwelcome howls of feedback and can prevent loudspeaker damage due to switch-on transients.
- Set all the channel EQs to their flat or neutral position and optimize the input gain control setting for each channel in turn using PFLs.
- If low frequency background noise is a problem, switch in the High Pass Filter on each of the microphone channels being used, except on low frequency sound sources such as basses and kick drums.
- Ring out the system as described above, with the vocal mics open, and notch out any obvious trouble spots.
- Establish the maximum working level for the lead vocal mic so as not to incur feedback and then work a little below this level to allow a margin of safety. Again, see the notes on ringing out the system.
- Set up the backing vocal mics and check that there is no feedback problem when both the backing vocal and lead vocal mics are on. If there is, reduce the master gain setting until the feedback disappears.
- Now the instrument and direct line inputs can be balanced relative to the vocals. Start with drums and work through to the bass and rythm instruments.
- Test out any effects units connected to the system and establish the correct balance of dry and effected sound.

Avoiding Feedback

- Turn down or mute any mics not in use. This reduces the risk of feedback and avoids the back line being picked up.
- If feedback is a real problem, consider moving the main PA speakers away from the mics a little. Also check the back of the stage, because if the wall is acoustically reflective, some sound from the room will be reflected back into the mics increasing the risk of feedback.
- Avoid excessive use of boosted EQ as this can encourage feedback and may also spoil the basic character of the sound. Consider it an aid to fine tuning rather than as a means of making radical changes.
- The use of stage monitors will also worsen the feedback situation so run these at the lowest volume that the performers can comfortably work with. Position the cabinets so as to allow as little direct sound as possible to enter the vocal microphones. If possible, use a graphic EQ on each monitor.

NB: Remember, people soak up sound! The perfect mix achieved in an empty venue will have to be tweaked when the crowds arrive. Sound waves are also affected by heat and humidity.

B. Larger Performances

Although the example shown in the 'Mixing Live' diagram shown at the beginning of this section is of a small band, the principles are the same no matter the size of the live performance or venue. However, for larger PAs additional speakers, monitors, effects and processors may be required as well as slightly different positioning for each of these pieces of equipment. These additional requirements are outlined below:

Medium Sized Venues

The console used will require more input channels. For example it is likely you will want to mic up all of the drums, and there are also likely to be more instruments, backing singers and sound sources in general.

More monitor sends will also be required - a single monitor will not be enough for larger bands. The bass and drums will require a monitor between them. The vocalists will want a monitor each so they can hear themselves above the band.

More speaker outputs may be needed in larger venues so that all the audience can be reached, without there being "holes" in the amplified audio signal. It may be necessary to record the event. This will require additional level controlled stereo outputs or direct outs if a multitrack is being used.

NB: For simplicity, these diagrams do NOT show any outboard equipment.

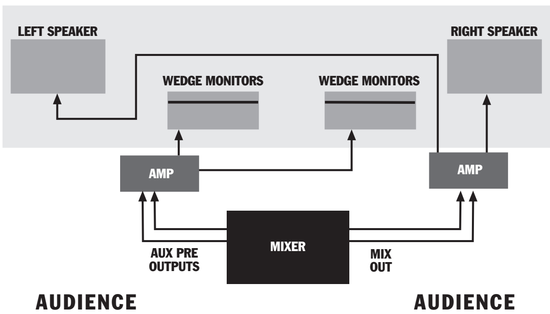
Large Sized Venues

Large venues will require a separate "Front of House" (FOH) console for the audience mix and a Monitor console for the band, as with a larger stage area each band member will require at least one monitor wedge. The auxiliary send system of the FOH console will not be able to cope with these demands alone as it will have to deal with several effects units.

The FOH console will have a large number of mic/line inputs, plus a large number of matrix outputs so that a complex range of speaker clusters can be placed around the auditorium.

SMALL VENUES

FIG. 4.2



MEDIUM SIZED VENUES

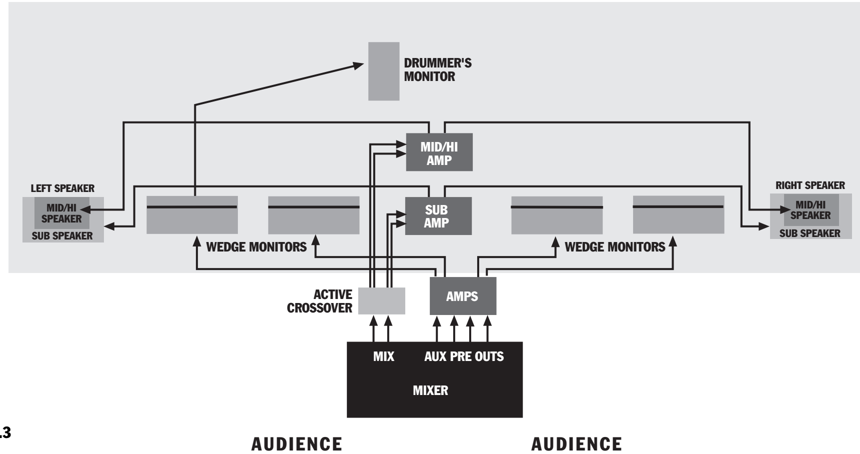


FIG. 4.3

LARGER VENUES

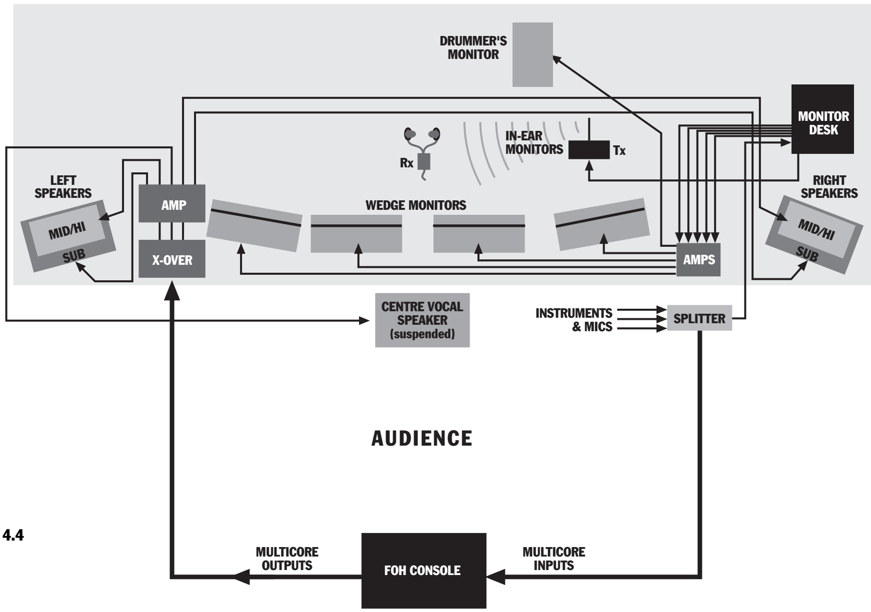


FIG. 4.4

C. Recording Live

In some situations, you may want to record a performance. Depending on the situation, the feed for recording may come from the FOH mixer, microphone splitter boxes, or your own microphones which have been set up alongside those of the band.

The diagram below shows a typical example of the sound sources being split between FOH and Recording. The recording console operates independently from the FOH mixer.

NB: When using Folio SX it will be necessary to re-patch for multitrack playback.

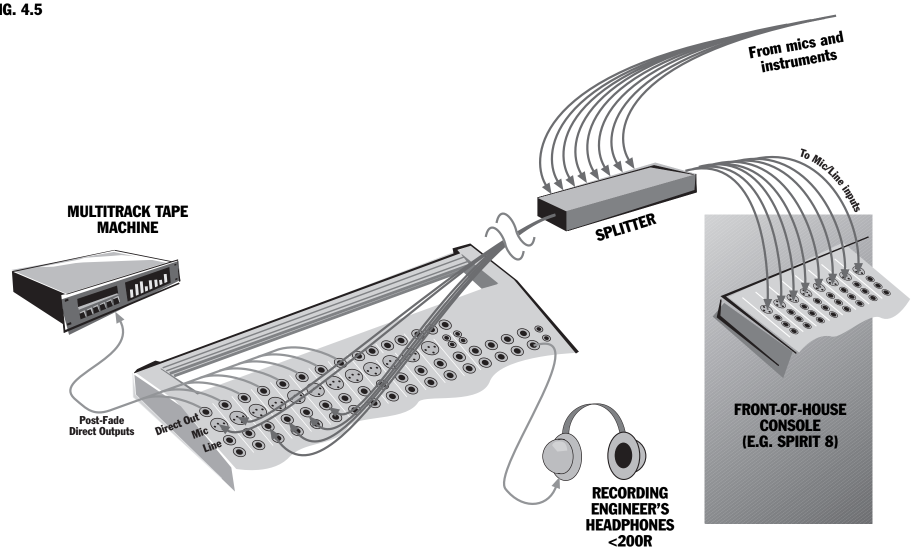
NB: Subgroups can be used for submixing many inputs (e.g. drums) to a multitrack input. This is useful when tape track availability is limited.

Hints & Tips

- Try to locate the mixer in a different room to the performance to avoid distraction from the live sound. If this is not possible, use a good pair of noise-excluding headphones for monitoring.
- Wherever possible, take feeds from mic splitters - this will provide clean, low-noise signals suitable for recording.
- Often, Tape Sends are unbalanced, so keep signal paths as short as possible between output and recorder to avoid interference.
- If there aren't enough microphones, use a stereo pair to pick up the overall sound and the rest to emphasize individual performers.
- Use a compressor/limiter to avoid overloading the digital input of the recorder.

RECORDING LIVE

FIG. 4.5



OTHER APPLICATIONS

A. Monitor Mixing

Monitors are used to allow band members to hear themselves.

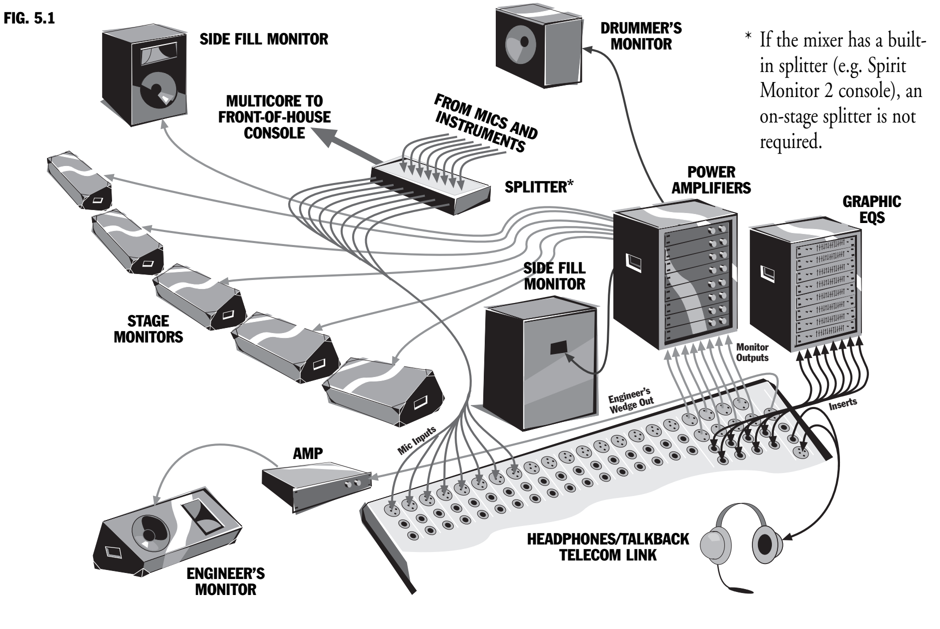
When dealing with the monitoring requirements of, say, a large live band, it is common practice to keep the monitor mix function totally separate from the Front of House console.

Some form of graphic equaliser in line with each monitor speaker is desirable as it allows troublesome frequencies to be notched out. The monitor system is rung out in exactly the same way as the main PA (see Ringing Out Section 4), and the final ringing out must be done with both the monitor and main PA systems set at their normal operating level.The monitoring console is situated off-stage and derives its feed direct from mic splitters. Note: the Spirit Monitor 2 console has its own built-in mic splitters.

- It is normal for a telecommunication link to be used between the FOH and monitor engineer so that they can talk to each other during the performance.
- Each stage monitor needs its own power amp. Keep things tidy by using rack-mounted stereo amps.
- Graphic EQs are patched via the console, like the power amps they should be rack-mounted for easy access.
- If the lead vocalist uses in-ear monitoring, he/she will be acoustically isolated, so it's a good idea to feed audience pick-up mics into his/her mix to provide a sense of involvement.
- 'Side fills' are often used where monitoring is required over a large stage area, floor space is at a premium, and too many wedge monitors would simply clutter things up both physically and acoustically. Don't compromise on these speakers - they'll have to work hard to punch sound through to the performers.
- The Monitor Engineer's wedge lets him hear the total foldback mix or selected parts thereof.
- A good Monitor Engineer, who is "invisible" to the audience, will always position himself so as to see visual signals from the performers.

MONITOR MIX

FIG. 5.1



* If the mixer has a built-in splitter (e.g. Spirit Monitor 2 console), an on-stage splitter is not required.

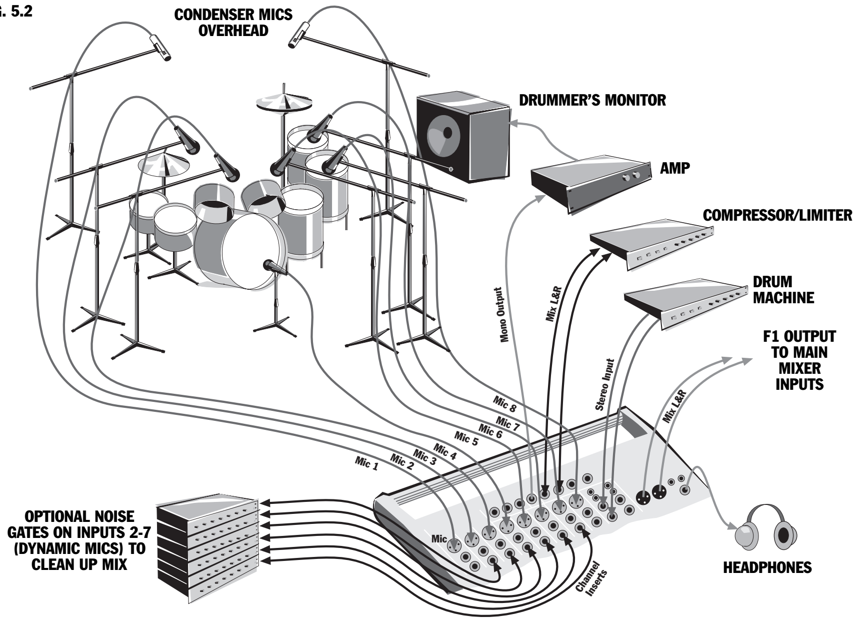
B. Submixing

There are certain groups of instruments or performers (drums, backing vocals, multi-keyboards, etc) that can be logically grouped together - to save on input channels - via a small mixer, the output of which can then be controlled by just one pair of faders on the master console.

- If a mono output is available it can be used for a drum fill or for recording purposes.
- Output from the submixer goes to the FOH console and/or may be used for a small recording set up.
- Use the Aux Returns on the FOH console to return the sub-mix. This saves valuable input channels on the FOH console.
- In the case of a drum kit where many mics are in close proximity, the use of Noise Gates will prevent spill and clean-up the mix.
- Use a Compressor/Limiter to maintain a consistent level.

SUBMIXING

FIG. 5.2



IN THE STUDIO

A. Essentials & Ergonomics

Think about room layout and equipment. No, we're not going to plan your studio for you, but here are a few pointers:

- If you play keyboards, set them up so that you can reach the mixer.
- Position your effects and synth modules within arms length.
- If you use a computer, position the screen so as to avoid reflections. Do not position speakers near the screen unless they are magnetically compensated or shielded.
- If the room is too 'live', deaden it with drapes or soft furnishings.
- For best results, use dedicated nearfield monitors.
- Don't use large speakers in a small room - they'll sound wrong at low frequencies.
- Do use a well specified power amp (minimum 50 watts per channel).
- Don't compromise on a weedy amp: it will distort at high levels and may damage the speakers.

B. Tape Machines & Recording Media

Basically, you'll need two types of tape machine: a **multitrack recorder** for recording the individual parts of the performance in readiness for mixdown onto a **2-track recorder** for mastering. There are both analogue and digital models available. The final choice must be based on individual requirements.

C. The Console

Studio work presents additional problems for a mixing console in that it has to deal with a two stage process requiring very different skills.

1 Recording - Sound sources have to be captured on multitrack tape. This process will include ensuring that the cleanest strongest signal is being recorded to tape, without overload and distortion, optimising the sound of the recorded signal with EQ, signal processing and effects, monitoring the recorded sources, and creating a headphone mix for the musicians to ensure the best possible performance from them.

2 Mixdown - All the recorded sound sources as well as any "live" media coming from sequencers, drum machines or samplers must then be blended together using EQ, level, pan and effects and mastered down to a two-track device to create a "final mix". This process bears some similarities to mixing a band - minus the audience, the live performance and poor venue acoustics!

If you have seen any T.V. shows including footage of commercial recording studios you may be forgiven for thinking that good multitrack recordings are only possible using a mammoth console. This does not have to be the case! Professional sounding results can be achieved, albeit with some repatching between recording and mixdown stages, using a relatively small multipurpose mixer. However, to achieve professional results the mixer must be equipped with either (and preferably both):

- **Direct outs**
- **Groups/Subs**

When purchasing a console for both live and recording work, ensuring these facilities are available will save you having to buy a dedicated recording console until your requirements become more sophisticted.

D. Simple Multitrack Recording

The diagram below shows a simple recording set-up using a multipurpose console equipped with direct outs and a pair of subgroups. The sound from instruments or voices is taken straight out to be recorded by the multitrack, with recorded signals being returned from the multitrack's channels into spare inputs of the mixer so they can be monitored. Alternatively, backing vocals or grouped instruments such as drumkits may be recorded to single or pairs of tracks by subgrouping them and connecting the mixer's group outputs to the multitrack device.

The engineer monitors both performances and previously recorded material through a monitor amp and speakers, with the performers getting their own separate foldback mix through the auxiliary sends .

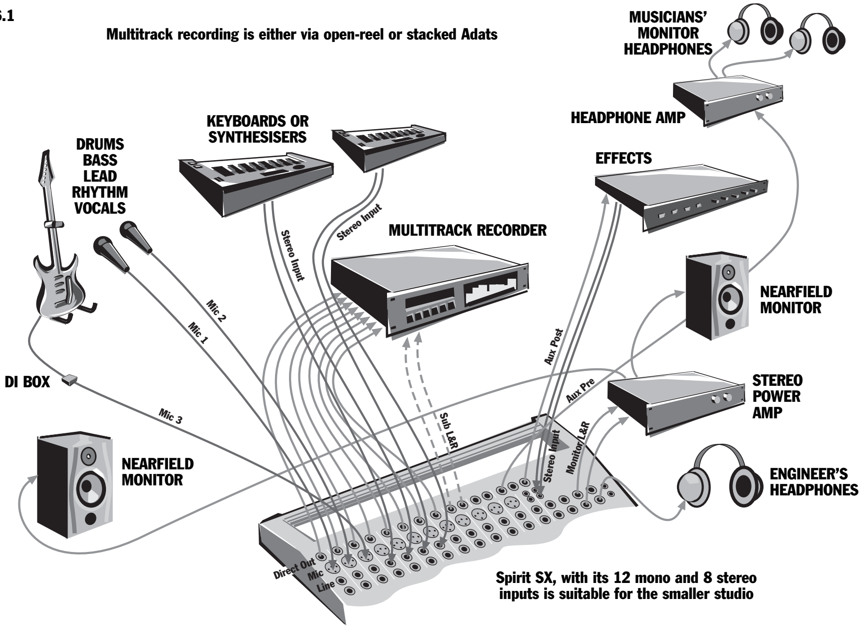
Hints and Tips when Recording:

- If you are recording as a solo performer on a budget, you can avoid the expense of buying a separate amp to create a headphone mix. Plug your headphones into the console's headphone connector and use its monitor mix for your foldback. Alter channel fader levels as you wish to achieve optimum headphone levels for your performance.
- If your console is not large enough to cope with every multitrack send and return, connect only as many Direct Outs as you need per take. For example, if you are recording solo you will only be recording one instrument at a time anyway, so a maximum of only two direct outs will be required for stereo instruments, and one for mono ones. The same channel direct outs may then be repatched to adjacent multitrack tape ins to record new tracks. This should leave enough channels free to monitor all your recorded tracks.
- If you run out of tape tracks, group instruments together. For example a fully mic'd up drumkit can be recorded in stereo to two tape tracks via a pair of groups, or if you are really stretched you could do this with the entire rhythm section, including bass and rhythm guitar. However, it is then essential to mix the balance between the instruments accurately as, once recorded, they can never be individually altered again.
- If you have only one effects unit and you need it to create a variety of different sounds, it may be neccessary to record the instrument with effects included. Again, remember that once you have done this there is no going back, so wherever possible it is best to record "dry" and buy a second effects unit if you can. If you must record "wet", look at you

MULTITRACK RECORDING

FIG. 6.1

Multitrack recording is either via open-reel or stacked Adats



Spirit SX, with its 12 mono and 8 stereo inputs is suitable for the smaller studio

mixer's block diagram and use outputs coming after the effects return for this purpose.

- Do not record in the same room in which you are playing unless your monitor speakers are muted. At the very least, your recorded track will pick up the mix from the monitor speakers, but more likely howl-round and feedback will occur which will damage your equipment. If you a recording a band, it is best to put them in an entirely different room altogether.
- Setting recording levels - for the best results, as it is important to set the highest record levels you can on your multitrack without getting overload or distortion. If you set levels too low, you will end up with a weak signal and background hiss. All multitrack recorders allow you to set record levels before a take. Consult the recorder's manual as to how best to achieve this.

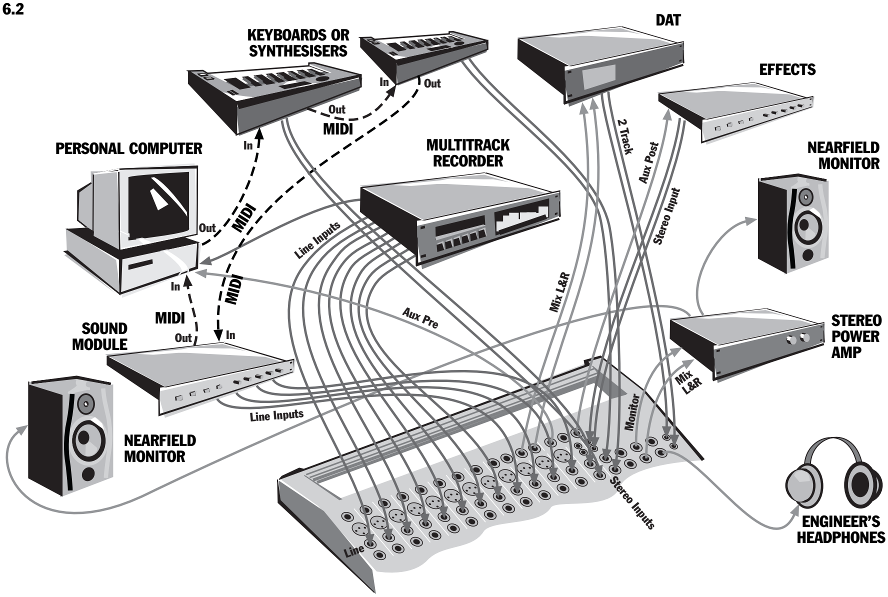
E. Simple Multitrack Mixdown

The diagram below shows how a simple set-up will look for the mixdown process. Some repatching has occured to free up the input channels which were used as mutitrack tape sends. Tape returns can then be plugged into the mixer in sequence from channel 1 upwards, leaving any spare inputs for sequenced MIDI instruments. Effects, amps and speakers may be left as before.

NB: Mixdown hints and tips may be found in "Creating a Mix" at the end of this section.

SIMPLE MULTITRACK MIXDOWN

FIG. 6.2



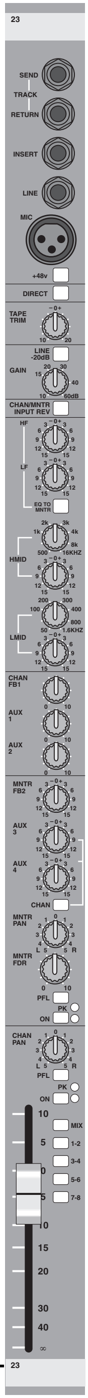
F. Using a Dedicated "In-Line" Mixing Console

For recording projects beyond 8 track, a multipurpose console is usually inadequate, being unable to cope with the additional multitrack sends and returns and with all the repatching that is required between recording and mixdown. In such cases, a dedicated "in-line" recording console is necessary. An example of the input strip of such a console is shown here.

Virtually all of the features and facilities are identical to a standard mixer - except one: As well as including full channel input facilities and a direct out (here called a tape send), the strip also includes an extra input for a multitrack tape return as well as some basic rotary level control and pan facilities for that input. This second input is known as the **Monitor Input** or **Monitor Return**. Using this technique allows a signal to and from a multitrack to be handled by one input strip, saving space and avoiding the confusion of having to find corresponding send and return signals in different areas of the console.

The major advantage of using an "in-line" recording console is that repatching is unnecessary. This is because both channel and tape return inputs can be swapped (using the switch marked "Chan/Mntr Input Rev"), giving the signal coming from multitrack all the EQ, Auxiliaries and the linear fader of the channel input for the mixdown process. This also leaves the monitor input free for sequenced MIDI gear such as keyboards. If more facilities are required for these sound sources, then EQ and auxiliaries may be shared between the two inputs.

With two inputs per channel, a 16 channel "in-line"console actually has 32 inputs available. This high input count and compactness has made "in -line" consoles extremely popular with project studios, programming and remixing suites and commercially successful bands' home studios. With prices tumbling all the time, "in-line" consoles are now barely more expensive than standard designs.



Multitrack Recording and Mixing with an "In-Line" Console

A more complex recording set up with an "in-line" console is shown opposite in Fig 6.4. Both multitrack ins and outs are plugged into the same channel strip, avoiding the need for repatching, whilst for sound proofing purposes, musicians are recorded in a separate room. Effects and signal processors are connected in an identical way to any other console via auxiliary sends and returns and insert points.

G. Recording Instruments and Voices

VOCALS

- Use a cardioid condenser mic positioned 9 inches (225mm) from the singer.
- A pop shield will reduce explosive 'p' and 't' sounds.
- If sibilance is a problem, change to a dynamic mic or move the singer back from the mic.

Recommended effects/processor settings:

EQ: Not normally required. But, if necessary, use the HPF (High Pass Filter) to reduce rumble.

Compressor: Attack as fast as possible;
Release around 0.5S, ratio between 4:1 and 8:1.

Reverb: Try a decay time of around 3 seconds and a pre-delay of 50mS.

DRUMS

- Place mics 2 inches (50mm) from the heads of snare and kick drum.
- For the kick drum, place the mic inside - pointing directly at where the beater strikes the drumhead.
- To fully mic a kit, use separate mics on all toms and hats.
- Use condenser mics 5ft (1.5m) overhead, spaced around 5ft (1.5m) apart, to pick up the entire drum sound, cymbals and "ambience".

Recommended effects/processor settings:

EQ: Boost at: 80Hz to add weight to kick drums, 6kHz to add sizzle to cymbals or edge to a snare. Cut at 250-300Hz to reduce boxiness on a kick drum or low toms.

Gate: Fast attack setting to allow percussive transients to pass through. Precise settings will depend on microphone type and placement.

Reverb: Keep kick drum 'dry'. Try a percussion plate setting with a 2.5S decay time on other drums.

FIG. 6.3

ELECTRIC GUITAR

- Some players prefer the sound of a valve amplifier, so be prepared to mic up the speaker cabinet using a cardioid dynamic mic.
- Experiment with mic positioning to achieve the desired sound.
- If preferred, the guitar can be DI'd via a recording preamp which incorporates an amp simulator.

Recommended effects/processor settings:

EQ: Boost at: 120Hz to add 'thump' to rock guitars, 2-3kHz to add bite, 5-7 kHz to add zing to clean rhythm sound. Cut at: 200-300Hz to reduce boxiness, 4kHz and above to reduce buzziness.

Compressor: Attack between 10 and 50mS; Release, around 0.3S, Ratio, between 4:1 and 12:1. Because of the noise generated by a typical electric guitar, use in conjunction with a gate or expander is advised.

Reverb: Plate or room, 1.5 to 4S; 30 to 60mS pre-delay.

ACOUSTIC GUITAR

- Use the best mic that you can, preferably a condenser type.
- For a natural tone, position the mic between 12-18ins from the guitar, aiming at where the neck joins the body.
- If recording in stereo, point a second mic towards the centre of the neck, about 12-18ins from the instrument.
- Acoustic guitars sound best in slightly live rooms, if necessary place a piece of acoustically reflective board beneath the player.

Recommended effects/processor settings:

EQ: Boost: between 5kHz and 10kHz to add sparkle. Cut between: 1kHz and 3kHz to reduce harshness, 100 and 200Hz to reduce boom. In busy pop mixes you can cut the low end to produce a more cutting rhythm sound.

Compressor: Attack 20 mS; Release, around 0.5S, Ratio, between 4:1 and 12:1.

Reverb: Bright setting such as Plate to add vitality. Decay time of between 2 to 3S.

BASS GUITAR

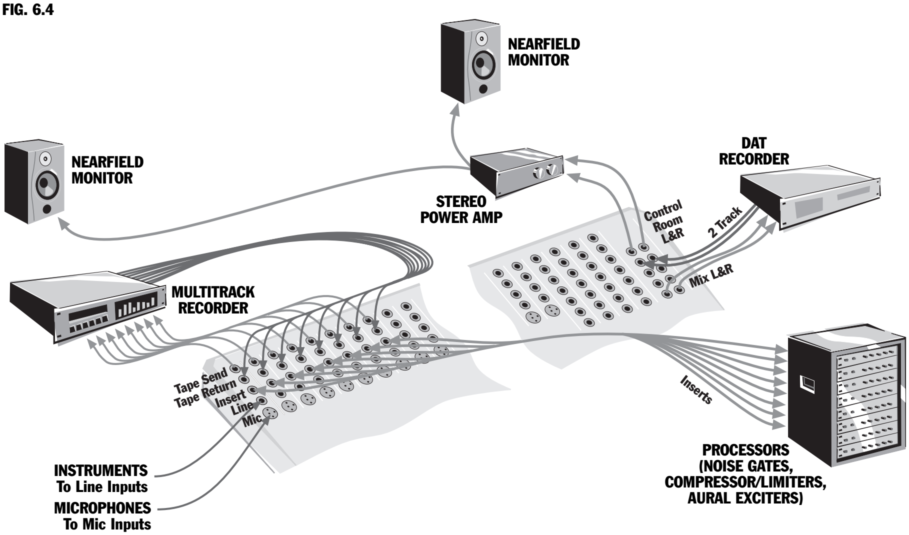
- Most engineers DI the bass via an active DI box and a compressor. This provides a clearer sound.
- Use the compressor to keep signal peaks under control.
- Check the player's technique; the harder the instrument is played, the brighter the tone.
- Consider the use of a budget graphic EQ.

Recommended effects/processor settings:

EQ: Boost: at 80-100Hz to add more weight and punch, between 2 and 4kHz to add edge. Cut: below 50Hz to

MULTITRACK RECORDING & MIXDOWN

FIG. 6.4



reduce unwanted rumble, between 180 and 250Hz to reduce boxiness.

Compressor: Attack around 50mS; Release, around 0.4S, Ratio, between 4:1 and 12:1.

KEYBOARDS

- Most electronic keyboards can be plugged directly into the line inputs of the mixing console.
- Bear in mind that the majority of contemporary synthesizers etc, have stereo outputs and will require two mixer channels.
- Most synthesizer sounds can be used without compression, though they do benefit from effects such as reverb or chorus.
- Overdriven keyboard sound may be created by feeding the signal via guitar recording preamp.

H. Planning a Session

- You have a lot to remember during a session, so create a track sheet to keep a log of what instrument is recorded onto what tape track, plus other relevant information.
- Record rhythm sections first; drums, bass, and rhythm guitar.
- Add vocals, solos, and additional instrumentation as overdubs.
- Decide whether you want to add effects at the mixing stage or while recording. If you can, try to keep a copy of the original "dry signal" on tape. You may wish to remix at a later date!
- When recording vocals, ask the singer what instruments they most need to hear in the headphone mix.

I. Creating a Mix

Go into 'neutral' before you start off -

- Set all the Aux Sends to zero.
- Set all EQ controls to their central positions.
- Pull all the faders down.
- All routing buttons 'up'.

Organize your Subgroups
- Put logical groups of sounds together.
- Route drums to a stereo sub-group.
- Consider grouping backing vocals.
- Group multiple keyboards.

Metering
- Use the PFL metering system for each channel in turn to optimize the gain setting.
- The PFL should just go into yellow band of the meter section, although peaking into the red area is acceptable.
- Check all the effects units for correct input levels.
- If fitted, use the Solo In Place function to check individual channels in isolation while retaining their original pan and level settings.

J. Balancing the Mix

If you don't have a lot of mixing experience, it can help to set up the drums and bass balance first, then move onto the vocals and the other instruments. Don't worry about fine tuning the EQ or effects until your dry mix is somewhere near right.

- Satisfy yourself that the mix is working in mono. Check for Phase problems.
- Pan bass drums, bass guitar and lead vocals to centre - this will stabilize the mix.
- Spread other instruments across the stereo stage as required, including backing singers.
- EQ the mix as required.
- Now add stereo effects as necessary to add to the illusion of space and width.
- Check the balance of your final mix by listening to it from the next room through the adjoining door: for some reason, this often shows up whether the vocals are too loud or quiet.

Hints & Tips

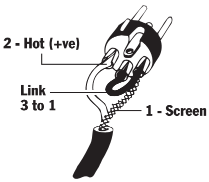
- Clean the heads of analogue tape machines before every session. Use cotton buds dipped in Isopropyl Alcohol.
- Check all instrument tunings before each take, because they have a tendency to change as the room warms up.
- Make a pop shield from stocking material stretched across a wire frame. This will minimise vocal "popping".
- Don't skimp on cables and connectors; these can be a source of noise.

WIRING & CONNECTORS

Faulty connectors and cabling are some of the most frequent sources of noise and poor sounding systems. The following section should help you connect your system correctly. It's also worth spending a little time referring to all of your user manuals, as wiring conventions can vary between manufacturers - see diagrams.

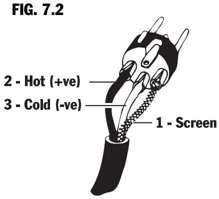
BALANCED AND UNBALANCED MIC INPUTS

FIG. 7.1



UNBALANCED INPUT

FIG. 7.2



BALANCED INPUT

Soundcraft uses XLR sockets for its balanced mic inputs. The wiring convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (–ve).

Balancing is a method of audio connection which cancels any interference in a signal, to give low noise operation. This is achieved by using a 2-conductor mic cable, usually surrounded by a shield, in which the 'hot' and 'cold' signals are opposite polarity. Any interference picked up will be of the same polarity on both hot and cold wires and will be rejected by the mic input's Difference Amplifier. You may use unbalanced sources when wired as shown. However, do not use unbalanced sources with Phantom Power switched on. The voltage on Pins 2 & 3 of the XLR connector may cause serious damage.

BALANCED AND UNBALANCED LINE INPUTS

Line inputs accept 'A' Gauge, 3-pole (Tip, Ring, Sleeve) 1/4 inch jack wired as shown in Fig. 7.3.

FIG. 7.3

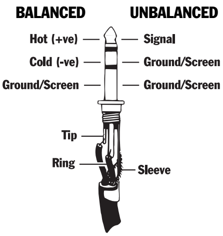
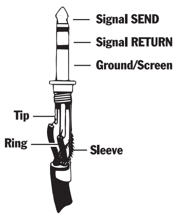


FIG. 7.4



Note that for unbalanced operation the screen of the cable is wired to both the Ring and the Sleeve of the jack.

INSERTS

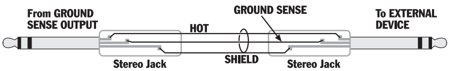
A Mixer insert point is a single, 'A' Gauge, 3-pole (stereo), switched jack socket (not unlike the headphone socket on a hi-fi amplifier). When a 3-pole jack is inserted the signal path is interrupted. The signal is taken out of the mixer via the plug tip, through an external piece of equipment and then back to the mixer on the ring of the plug. A special Y-cord is required which has the stereo jack at one end and two mono jacks, for the processor's input and output, at the other. See Fig. 7.4.

GROUND COMPENSATED OUTPUTS

Ground compensated outputs may, to all intents and purposes, be treated as balanced outputs. Ground compensation helps avoid hum loops when the console is feeding into an unbalanced piece of equipment. Essentially, the Ground Compensated output has three connections, much like a conventional balanced output, except that the pin normally designated 'cold' acts as a 'ground sense' line enabling it to sense and cancel any ground hum present at the output.

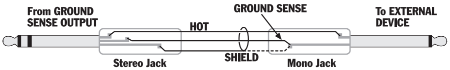
BALANCED CONNECTION

FIG. 7.5



UNBALANCED CONNECTION

FIG. 7.6



The convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot, Pin 3 - Ground Sense. For jacks, the wiring convention is: Tip - Hot, Ring - Ground Sense, Sleeve - Shield.

For use with balanced destinations, the Ground Sense output may be treated as 'cold' allowing the connection to be made normally. Where the destination has an unbalanced jack input, a two-core (balanced-type) lead should be made up as shown. Unbalanced jacks may also be plugged directly into Ground Compensated Output jack sockets, but the benefit of hum rejection will be lost.

IMPEDANCE BALANCED OUTPUTS

Impedance Balanced Outputs are configured as normal balanced outputs: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (–ve). See Fig. 7.2.

Impedance Balanced Outputs work on the principle that hot and cold terminals have the same resistance. When impedance balanced outputs are used with a balanced input, good rejection is achieved for both common-mode ground voltages and electrostatic interference.

Note: The cold terminal can be either shorted to ground locally or left open-circuit for balanced and unbalanced operation.

GLOSSARY

ACOUSTIC FEEDBACK (HOWLROUND)
A whistling or howling noise caused by an amplified signal 'feeding back' into the amplification chain via a microphone or guitar pick-up.

ACTIVE DI BOX
A device which permits Direct Injection of signals from guitars, etc, into the console. Incorporates circuitry to adjust gain and provide impedance matching. Requires power and may be battery driven or sometimes 'phantom powered' from a console.

AFL (AFTER FADE LISTEN)
A function that allows the operator to monitor a post-fade signal. Used with Aux Masters.

AMPLIFIER
Device that increases the level of an electrical signal.

AMPLITUDE
Signal level, usually in volts.

ANALOGUE
Analogy (n.): correspondence or partial similarity, using physical variables. For example; an analogue tape recorder stores sound on tape in the form of a magnetic pattern which is a replica of the original musical waveform.

ASSIGN
On a mixing console, to switch or route a signal to a particular signal path or combination of signal paths.

ATTENUATE
To decrease the level of a signal.

AUXILIARY SEND
Level control feeding a dedicated bus for driving external effects or a foldback monitoring system. An output from the console comprising a mix of signals from channels derived independently of the main stereo/group mixes. Typically the feeds to the mix are implemented on rotary level controls.

BACK-LINE
Stage parlance for the row of instrument amplifiers and loudspeaker cabinets behind the performers, e.g. guitar amps.

BALANCE
Relative level of the left and right channels of a stereo signal.

BALANCED
A method of audio connection which 'balances' the signal between two wires and a screen which carries no signal. Any interference is picked up equally by the two wires, through common mode rejection at the destination differential balanced input resulting in cancellation of the interference signal. For balancing to be effective, both the sending and receiving device must have balanced output and input stages respectively.

BANDWIDTH
A means of specifying the range of frequencies passed by an electronic device such as an amplifier, mixer or filter.

BARGRAPH
A row of LEDs calibrated to indicate signal level.

BOOST/CUT CONTROL
A single EQ control which allows the range of frequencies passing through its filter to be either amplified or attenuated. The centre position is usually the 'flat' or 'neutral' position.

BUS or BUSS
A defined set of conductors along which signals may travel. A mixer has several busses carrying the stereo mix, the groups, the PFL signal, the aux sends, etc.

CAPACITOR
See Condenser

CARDIOID PATTERN
The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front. Mainly used for stage vocals or in any situation were sound has to be picked up from a concentrated area, i.e. drums.

CHANNEL
A strip of controls in a mixing console relating to a single mono input or a stereo input.

CHIP
Integrated circuit; a multi-pinned device consisting of many circuits encapsulated in plastic.

CHORUS
Effect created by doubling a signal and adding delay and pitch modulation.

CLIPPING
Severe form of audio distortion which is the result of signal peaks exceeding the amplifier capacity. Normally caused by a limitation of the unit's power supply.

CLONE
Exact duplicate. Often refers to digital copies of digital tapes.

CONDENSER
Electrical component exhibiting capacitance (the ability to temporarily store electric current) and block direct current.

CONDENSER MICROPHONE
A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. Condenser mics are very sensitive, especially to distant sounds and high frequencies. They have to be powered, which can be achieved by batteries, but for professional use a 48v DC PHANTOM POWER supply is provided from the console via the balanced mic cable.

CONDUCTOR
A thing that conducts or transmits heat or electricity.

COMPRESSOR
A device designed to control or reduce the dynamic range of an audio signal.

CROSSOVER
A passive circuit, normally built into a speaker system which divides the full-range audio signal from an amplifier in order to feed the individual drive units, ie: bass, midrange and treble.

CUEING
To put a piece of equipment in readiness to play a particular part of the recording material. Assisted on a mixing console by use of the PFL (Pre-Fade LIsten) facility.

CUT-OFF FREQUENCY
The frequency at which the gain of an amplifier or filter has fallen by 3dB.

DAT (DIGITAL AUDIO TAPE)
High quality cassette based recording format which stores signals digitally and therefore provides very high quality sound. Originally touted for consumer use, but now firmly ensconced as a professional tool.

dB (DECIBEL)
A ratio of two signal levels. Can be in Voltage, Watts or Current units.

dBm
Variation on dB referenced to 0dB = 1mW into 600 ohms.

dBu
Variation on dB referenced to 0dB = 0.775 volts.

dBV
Variation on dB referenced to 0dB = 1 Volt.

DETENT
In audio terms a click-stop in the travel of a rotary or slide control, normally used to indicate 'centre stereo' on pan-pots or 'zero boost/cut' on EQ controls.

DI BOX
A device allowing connections as explained below.

DI (DIRECT INJECTION)
The practice of connecting an electric musical instrument directly to the input of the mixing console, rather than to an amplifier and loudspeaker which is covered by a microphone feeding the console.

DIGITAL DELAY
The creation of delay and echo effects in the digital domain.The premise being that, as digital signals are resistant to corruption, the process will not introduce additional noise or distortion.

DIGITAL REVERB
Reverberation effects created as above.

DIGITAL
The processing and storage of signals with sound- information represented in a series of '1s' and '0s', or binary digits.

DIRECT OUTPUT
A pre-/post-fade, post-EQ line level output from the input channel, bypassing the summing amplifiers, typically for sending to individual tape tracks during recording.

DRY
Slang term for an original audio signal that has had no added effects.

DYNAMIC RANGE
The ratio in decibels between the quietest and loudest sounds in the audible range that the audio equipment will reproduce.

DYNAMIC MICROPHONE
A type which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

EARTH
See GROUND.

EFFECTS
The use of devices to alter or process the sound to add special effects eg; reverb, normally as a mix of the original ('dry') sound and the treated ('wet') version.

EFFECTS RETURN
Additional mixer input designed to accommodate the output from an effects unit.

EFFECTS LOOP
Connection system that allows an external signal processor to be connected into the audio chain.

EFFECTS SEND
A post-fade auxiliary output used to add effects to a mix.

ELECTRET MICROPHONE
Type of condenser microphone using a permanently charged capsule.

ELECTRONIC CROSSOVER
An active device which divides the full range audio signal into several narrower frequency bands (eg:low, mid and high), which are then amplified and fed to the appropriate speaker drive units.

ENCLOSED HEADPHONES
Types that completely enclose the ears and therefore provide good exclusion of outside noise. Of particular use when monitor mixing or recording live on stage.

EQ
Abbreviation for equaliser or equalisation.

EQUALISER
A device that allows the boosting or cutting of selected bands of frequencies in the signal path.

EXPANDER
The opposite of a compressor, an expander increases the dynamic range of signals falling below a pre-determined threshold.

FADER
A linear control providing level adjustment. Favoured by professionals due to smoothness of activation and the ability to give an instant visual indication of status.

FILTER
A filter is a device or network for separating waves on the basis of their frequencies.

FOH
An acronym for Front Of House. In the entertainment world "House" is a collective term for the audience at a theatre, cinema, etc. Hence an FOH console will be situated "audience-side" of the stage. A "house" PA system refers to the main audio system responsible for the principal sound in the venue.

FOLDBACK
A feed sent back to the artistes via loudspeakers or headphones to enable them to monitor the sounds they are producing.

FOLDBACK SEND
A pre-fade auxiliary output used to set up an independent monitor mix for the performers.

FREQUENCY RESPONSE
The variation in gain of a device with frequency.

FSK (Frequency Shift Keying)
A method of synchronisation which generates a series of electronic tones related to the tempo of the music. These tones may then be record on a spare track of the multitrack recorder.

FX UNIT
Slang term for Effects Unit. Typical effects units are delays, reverbs, pitch shifters, and chorus units.

GAIN
Gain is the factor of how much the level of a signal is increased or amplified. Normally expressed in decibels.

GATE
A user-adjustable electronic device that switches off the signal path when the signal falls below a certain predetermined level or threshold.

Typically used to ensure silence between pauses in the signal during vocal passages or to prevent ‘spill’ between the close-proximity, multiple mics on a drum kit.

GRAPHIC EQUALISER
Device incorporating multiple narrow-band circuits allowing boost and cut of predetermined frequencies. Vertical fader controls are used which provide a ‘graphic’ representation of the adjustments across the frequency range.

GROUND COMPENSATION
A technique used to cancel out the effect of ground loops caused by connections to external equipment.

GROUND
Ground and Earth are often assumed to be the same thing, but they are not. Earth is for electrical safety, while Ground is the point of zero voltage in a circuit or system.

GROUND LOOP
A ground loop occurs when there are too many ground points, allowing small electrical currents to flow.

GROUP
An output into which a group of signals can be mixed.

HEADROOM
The available signal range above the nominal level before clipping occurs.

HERTZ (Hz)
Cycles (or vibrations) per second.

HIGH PASS FILTER
A filter that rejects low frequencies below a set frequency, typically 100Hz.

IMPEDANCE
The AC resistance of a circuit which has both resistive and reactive components.

IMPEDANCE BALANCING
A technique used to minimise the effect of hum and interference when connecting to external balanced inputs.

INDUCTOR
Reactive component that presents an increasing impedance with frequency. A coil in a loudspeaker crossover is an inductor.

INSERT POINT
A break point in the signal path to allow the connection of external devices, for example signal processors or to another mixer.

K OHM, K Ω or kHz
x 1000 ohms, x 1000 ohms and x 1000Hz respectively.

LINE LEVEL
Signals at a nominal level of -10dBV to +4dBu, usually coming from a low impedance source such as keyboards, drum machines,etc.

mA (milliampere)
One thousandth of an ampere, a measure for small electrical currents.

MIC SPLITTER
A device which divides the output from a microphone in order to supply two signals, for example; FOH console and recording mixer or monitor console.

MIDBAND
The range of frequencies to which the human ear is most sensitive.

MIDI
Musical Instrument Digital Interface.

MIXDOWN
The process of taking the outputs from a multitrack recorder, processing as required and combining all elements to create a stereo ‘master’.

MONITOR LOUDSPEAKER
Any high quality loudspeaker which is used to check the quality or status of the signal.

MTC (MIDI Time Code)
An interpretation of SMPTE allowing the time code to come in as part of the MIDI data stream.

MULTICORE
A cable with multiple cores allowing signals to be carried independently but within the same physical outer casing.

MUTE GROUPS
A method of combining the on/off status of a selection of channels under a single control button.

NEARFIELD MONITOR
A high quality, compact loudspeaker designed for use at a distance of three to four feet from the operator. Their use ensures that detrimental room effects are minimised.

NORMALISE
A socket is said to be normalised when it is wired in such a way that the original signal path is maintained unless a plug is inserted into the socket. The most common examples of normalised connectors are the INSERT POINTS found on mixing consoles.

OSCILLATOR
A tone generator for test and line-up purposes.

OVERDUB
To add another part to a multitrack recording or replace one of the existing parts.

OVERLOAD
To exceed the operating capacity of an electronic or electrical circuit.

PAN (POT)
Abbreviation of 'panorama': controls levels sent to left and right outputs. Allows positioning of signals within the stereo sound stage.

PARAMETRIC EQUALISER
A graphic equaliser in which the cut/boost, frequency and bandwidth are all adjustable.

PASSIVE
A circuit or component which does not amplify the signal or is not powered.

PATCH BAY
A system of panel mounted connectors used to bring inputs and outputs to a central point from where they can be routed using plug-in patch cords.

PATCH CORD
Short cable used with patch bays.

PEAKING
A signal of the maximum displacement from its mean (average) position.

PHANTOM POWER
The +48v DC voltage applied equally to the two signal pins of a balanced mic input to provide powering for condenser microphones.

PHASE
Phase is the fraction of the whole period that has elapsed, measured from a fixed datum. A term used to describe the relationship of two audio signals: in-phase signals reinforce each other, out-of-phase signals result in cancellation.

PHONO PLUG
A hi-fi connector developed by RCA and used extensively on semi-pro recording equipment.

POLARITY
The orientation of the positive and negative poles of an audio connection. Normally, connections are made positive to positive, negative to negative and this would ensure correct polarity. If this is reversed the result will be out-of-phase signals (see PHASE above).

POP SHIELD
A device used in the studio, consisting of a thin mesh placed between the microphone and vocalist in order to reduce the 'explosive' effects of 'P' and 'T' sound

POST-FADE
The point in the signal path after the channel or master fader and therefore affected by fader position.

PRE-FADE LISTEN (PFL)
A function that allows the operator to monitor the pre-fade signal in a channel before it reaches the main mix.

PRE-FADE
The point in the signal path before the monitor or master position and therefore unaffected by the fader setting.

PROCESSOR
A device which affects the whole of the signal passing through it, e.g. gate, compressor or equaliser.

Q (Bandwidth)
A measure of the sharpness of a bandpass filter. The higher the value of Q, the narrower the band of frequencies that passes through the filter.

RESISTANCE
Opposition to the flow of electrical current.

REVERB
Acoustic ambience created by multiple reflections in a confined space. A diffuse, continuously smooth decay of sound.

RINGING OUT
The process of finding the problem frequencies in a room by steadily increasing the gain of the system until feedback occurs. A GRAPHIC EQUALISER is then used to reduce the offending frequencies.

ROLL-OFF
A fall in gain at the extremes of the frequency response. The rate at which a filter attenuates a signal once it has passed the filter cut-off point.

SEQUENCER
Computer-based system for the recording, editing and replay of MIDI music compositions.

SHELVING
An equaliser response affecting all frequencies above or below the break frequency i.e. a high-pass or low-pass derived response.

SHORT CIRCUIT
The situation where two electrical conductors touch.

SIBILANCE
n. sounding with a hiss. When certain phonics are exaggerated, ie: s, sh.

SIGNAL
Electrical representation of input such as sound.

SIGNAL CHAIN
The route taken by a signal from the input to a system through to its output.

SIGNAL-TO-NOISE RATIO
An expression of the difference in level between the audio signal and the background noise of the device or system. Normally expressed in decibels.

SMPTE (Society of Motion Picture and Television Engineers)
Time code developed for the film industry but now extensively used in music and recording. SMPTE is a real-time code and is related to hours, minutes, seconds and film or video frames rather than to musical tempo.

SOLO-IN-PLACE
A function that allows the operator to listen to a selected channel on it's own but complete with all relevant effects, by automatically muting all other inputs.

SOUNDCRAFT
The name found on some of the best-value professional audio equipment around.

SOUND REINFORCEMENT
The process of amplifying or reinforcing on-stage sound (whether from already-amplified or acoustic instruments/voices) without overpowering the original sound. Suitable for smaller venues and often used solely to raise the level of the vocals above the back line and drums.

SPL (Sound Pressure Level)
Intensity of sound measured in decibels.

STEREO
Two channel system feeding left and right speakers to create the illusion of a continuous sound field. Stereo: from the Greek word for 'solid'.

STEREO RETURN
An input designed to receive any stereo line level source such as the output of effects or other external processing devices.

STRIPE
To record time code onto one track of a multitrack tape machine.

SWEEP EQ
An equaliser section (e.g. Midband EQ) which boosts or cuts a variable rather than fixed frequency.

TALKBACK
A system allowing the operator to speak to the artistes or to tape via the auxiliary or group outputs.

TAPE RETURN
A line level input provided specifically to receive the playback output of a tape machine.

TRANSIENT
An instantaneous rise in the signal level e.g. a cymbal crash or similar.

TRIM CONTROL
A variable control which gives adjustment of signal level over a limited and predetermined range usually for calibration purposes.

TRS JACKS
A 3-pole jack with Tip, Ring and Sleeve connection. Sometimes referred to as a stereo or A-gauge jack plug.

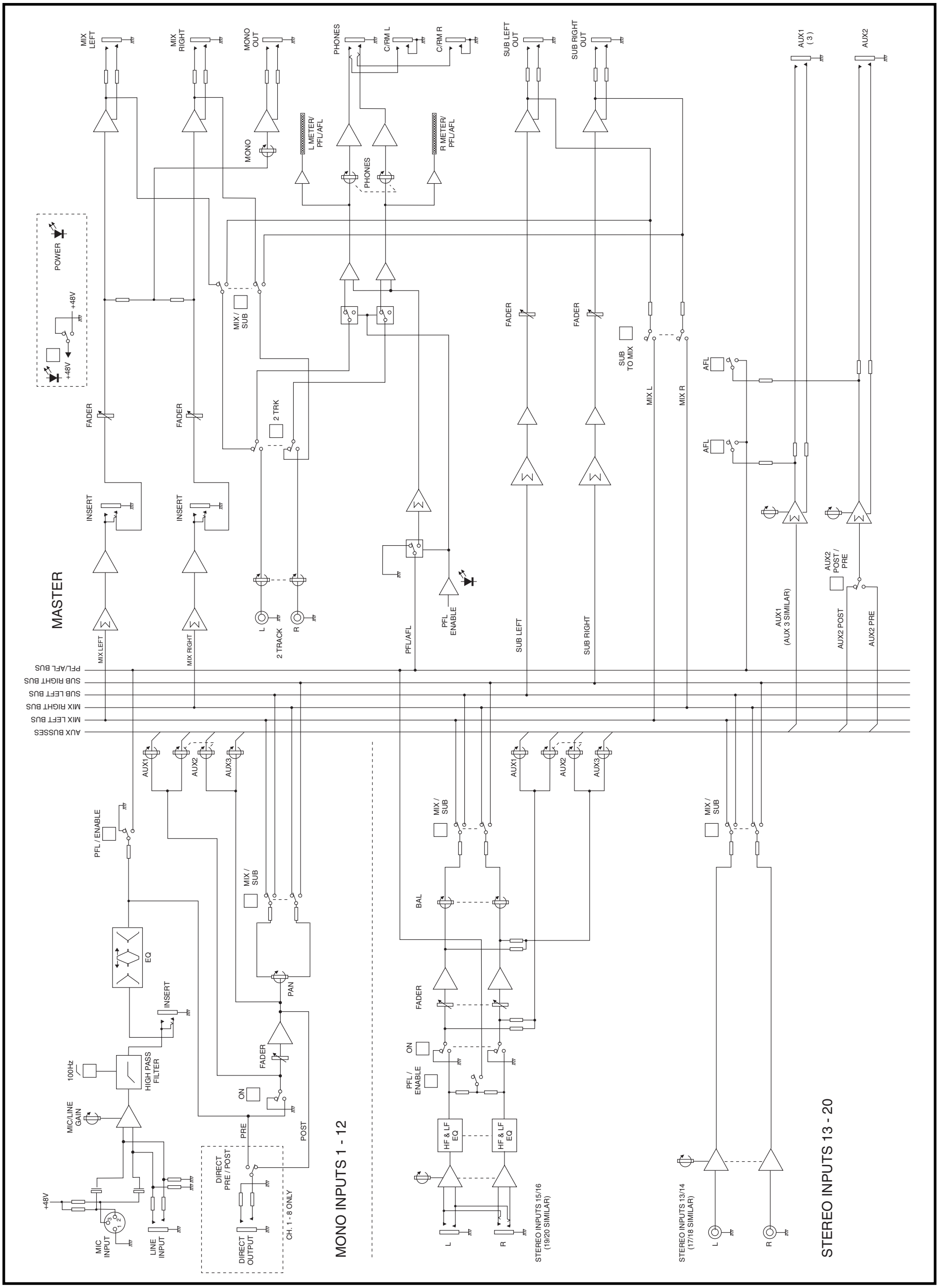
UNBALANCED
A method of audio connection which uses a single signal wire and the cable screen as the signal return.This method does not provide the same degree of noise immunity as a BALANCED connection.

WET
Slang term for a signal with added effects such as REVERBERATION, ECHO, DELAY or CHORUS.

Y-LEAD
A lead split so that one source can feed two destinations. Y-leads may also be used in console insert points in which case a stereo jack plug at one end of the lead is split into two mono jacks at the other.

2-TRACK RETURN
A line level stereo input on a mixing console designed to accept the output from a 2-Track recording device. May also be used as an additional effects return, depending on the internal routing of the mixer.

A Typical Block Diagram (Spirit SX shown)



How to Achieve

Depth and Dimension

in Recording, Mixing and Mastering

Introduction: Master the 2-channel art first. We can make much better 2-channel recordings than are common...

Everyone is talking about multichannel sound. I have no doubt that well-engineered multi-channel recordings will produce a more natural soundfield than we've been able to achieve in our 2-channel recordings, but it amazes me how few engineers really know how to take advantage of good ol' fashioned 2-channel stereo. I've been making "naturalistic" 2-channel recordings for many years, and there are others working in the pop field who produce 2-channel (pop, jazz, even rock) recordings with beautiful depth and space. I'm rather disappointed in the sound of 2-channel recordings made by simple "pan-potted mono", the typical sound of a rock mix. But it doesn't have to be, if you study the works of the masters.

I wonder if the recording engineers who are disappointed in 2-channel recording may simply be using the wrong techniques. Pan-potted mono techniques, coupled by artificial reverberation---tend to produce a vague, undefined image, and I can understand why many engineers complain about how difficult it is to get definition working in only two channels. They say that when they move to multichannel mixing (e.g., 5.1) that they have a much easier time of it. Granted, though I suggest that first they study how to make a good 2-channel mixdown with depth, space, clarity, and definition. It's possible if you know the tricks. Most of those tricks involve the use of the Haas effect, phase delays, more natural reverbs and unmasking techniques. If engineers don't study the art of creating good 2-channel recordings, when we move to 5.1, ultimately we will end up with more humdrum mixes, more "pan-potted mono", only with more speakers. This article describes techniques that will help you with 2-channel and multichannel recordings. Furthermore, well-engineered 2-channel recordings have encoded ambience information which can be extracted to multichannel, and it pays to learn about these techniques.

The Perception of Depth

At first thought, it may seem that depth in a recording is achieved by increasing the ratio of reverberant to direct sound. But it is a much more involved process. Our binaural hearing apparatus is largely responsible for the perception of depth. But recording engineers were concerned with achieving depth even in the days of monophonic sound. In the monophonic days, many halls for orchestral recording were deader than those of today. Why do monophonic recording and dead rooms seem to go well together? The answer is involved in two principles that work hand in hand: 1)The masking principle and 2)The Haas effect.

The Masking Principle

The masking principle says that a louder sound will tend to cover (mask) a softer sound, especially if the two sounds lie in the same frequency range. If these two sounds happen to be the direct sound from a musical instrument and the reverberation from that instrument, then the initial reverberation can appear to be covered by the direct sound. When the direct sound ceases, the reverberant hangover is finally perceived.

In concert halls, our two ears sense reverberation as coming diffusely from all around us, and the direct sound as having a distinct single location. Thus, in halls, the masking effect is somewhat reduced by the ears' ability to sense direction.

In monophonic recording, the reverberation is reproduced from the same source speaker as the direct sound, and so we may perceive the room as deader than it really is, because of directional masking. Furthermore, if we choose a recording hall that is very live, then the reverberation will tend to intrude on our perception of the direct sound, since both will be reproduced from the same location-the single speaker. So there is a limit to how much reverberation can be used in mono.

This is one explanation for the incompatibility of many stereophonic recordings with monophonic reproduction. The larger amount of reverberation tolerable in stereo becomes less acceptable in mono due to directional masking. As we extend our recording techniques to 2-channel (and eventually multichannel) we can overcome directional masking by spreading reverberation spatially away from the direct source, achieving both a clear (intelligible) and warm recording at the same time.

The Haas Effect

The Haas effect can be used to overcome directional masking. Haas says that, in general, echoes occurring within approximately 40 milliseconds of the direct sound become fused with the direct sound. We say that the echo becomes "one" with the direct sound, and only a loudness enhancement occurs.

A very important corollary to the Haas effect says that fusion (and loudness enhancement) will occur even if the closely-timed echo comes from a different direction than the original source. However, the brain will continue to recognize (binaurally) the location of the original sound as the proper direction of the source. The Haas effect allows nearby echoes (up to approximately 40 ms. delay, typically 30 ms.) to enhance an original sound without confusing its directionality. We can take advantage of the Haas effect to naturally and effectively convert an existing 2-channel recording to a 4-channel or surround medium. When remixing, place a discrete delay in the surround speakers to enhance and extract the original ambience from a previously recorded source! No artificial reverberator is needed if there is sufficient reverberation in the original source. Here's how it works:

Because of the Haas effect, the ear fuses the delayed with the original sound, and still perceives the direct sound as coming from the front speakers. But this does not apply to ambience-ambience will be spread, diffused between the location of the original sound and the delay (in the surround speakers). Thus, the Haas effect only works for correlated material; uncorrelated material (such as natural reverberation) is extracted, enhanced, and spread directionally. Dolby laboratories calls this effect "the magic surround", for they discovered that natural reverberation was extracted to the rear speakers when a delay was applied to them. Dolby also uses an L minus R matrix to further enhance the separation. The wider the bandwidth of the surround system and the more diffuse its character, the more effective the psychoacoustic extraction of ambience to the surround speakers.

There's more to Haas than this simple explanation. To become proficient in using Haas in mixing, study the original papers on the various fusion effects at different delay and amplitude ratios.

Haas' Relationship To Natural Environments

We may say that the shorter echoes which occur in a natural environment (from nearby wall and floor) are correlated with the original sound, as they have a direct relationship. The longer reverberation is uncorrelated; it is what we call the ambience of a room. Most dead recording studios have little or no ambient field, and the deadest studios have only a few perceptible early reflections to support and enhance the original sound.

In a good stereo recording, the early correlated room reflections are captured with their correct placement; they support the original sound, help us locate the sound source as to distance and do not interfere with left-right orientation. The later uncorrelated reflections, which we call reverberation, naturally contribute to the perception of distance, but because they are uncorrelated with the original source the reverberation does not help us locate the original source in space. This fact explains why the multitrack mixing engineer discovers that adding artificial reverberation to a dry, single-miked instrument may deteriorate the sense of location of that instrument. If the recording engineer uses stereophonic miking techniques and a liver room instead, capturing early reflections on two tracks of the multitrack, the remix engineer will need less artificial reverberation and what little he adds can be done convincingly.

Using Frequency Response to Simulate Depth

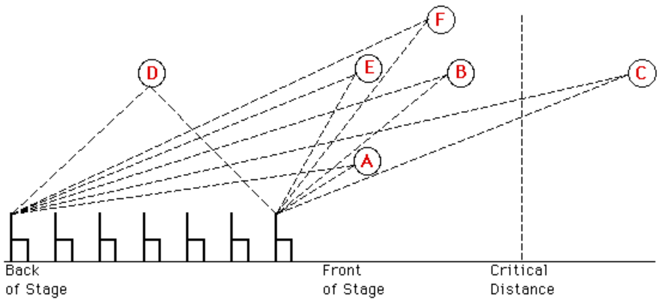
Another contributor to the sense of distance in a natural acoustic environment is the absorption qualities of air. As the distance from a sound source increases, the apparent high frequency response is reduced. This provides another tool which the recording engineer can use to simulate distance, as our ears have been trained to associate distance with high-frequency rolloff. An interesting experiment is to alter a treble control while playing back a good orchestral recording. Notice how the apparent front-to-back depth of the orchestra changes considerably as you manipulate the high frequencies.

Recording Techniques to Achieve Front-To-Back Depth

Minimalist Techniques

Balancing the Orchestra. A musical group is shown in a hall cross section. Various microphone positions are indicated by letters A-F.

Microphones A are located very close to the front of the orchestra. As a result, the ratio of A's distance from the back compared to the front is very large. Consequently, the front of the orchestra will be much louder in comparison to the rear. Front-to-back balance will be exaggerated. However, there is much to be said in favor of mike position A, since the conductor usually stands there, and he purposely places the softer instruments (strings) in the front, and the louder (brass and percussion) in the back, somewhat compensating for the level discrepancy due to location. Also, the radiation characteristics of the horns of trumpets and trombones help them to overcome distance. These instruments frequently sound closer than other instruments located at the same physical distance because the focus of the horn increases direct to reflected ratio. Notice that orchestral brass often seem much closer than the percussion, though they are placed at similar distances. You should take these factors into account when arranging an ensemble for recording. Clearly, we also perceive depth by the larger ratio of reflected to direct sound for the back instruments.



The farther back we move in the hall, the smaller the ratio of back-to-front distance, and the front instruments have less advantage over the rear. At position B, the brass and percussion are only two times the distance from the mikes as the strings. This (according to theory) makes the back of the orchestra 6 dB down compared to the front, but much less than 6 dB in a reverberant hall , because level changes less with distance.

For example, in position C, the microphones are beyond the critical distance---the point where direct and reverberant sound are equal. If the front of the orchestra seems too loud at B, position C will not solve the problem; it will have similar front-back balance but be more buried in reverberation.

Using Microphone Height To Control Depth And Reverberation

Changing the microphone's height allows us to alter the front-to-back perspective independently of reverberation. Position D has no front-to-back depth, since the mikes are directly over the center of the orchestra. Position E is the same distance from the orchestra as A, but being much higher, the relative back-to-front ratio is much less. At E we may find the ideal depth perspective and a good level balance between the front and rear instruments. If even less front-to-back depth is desired, then F may be the solution, although with more overall reverberation and at a greater distance. Or we can try a position higher than E, with less reverb than F.

Directivity Of Musical Instruments

Frequently, the higher up we move, the more high frequencies we perceive, especially from the strings. This is because the high frequencies of many instruments (particularly violins and violas) radiate upward rather than forward. The high frequency factor adds more complexity to the problem, since it has been noted that treble response affects the apparent distance of a source. Note that when the mike moves past the critical distance in the hall, we may not hear significant changes in high frequency response when height is changed.

The recording engineer should be aware of how all the above factors affect the depth picture so he can make an intelligent decision on the mike position to try next. The difference between a B+ recording and an A+ recording can be a matter of inches. Hopefully you will recognize the right position when you've found it.

Beyond Minimalist Recording

The engineer/producer often desires additional warmth, ambience, or distance after finding the mike position that achieves the perfect instrumental balance. In this case, moving the mikes back into the reverberant field cannot be the solution. Another call for increased ambience is when the hall is a bit dry. In either case, trucking the entire ensemble to another hall may be tempting, but is not always the most practical solution.

The minimalist approach is to change the microphone pattern(s) to less directional (e.g., omni or figure-8). But this can get complex, as each pattern demands its own spacing and angle. Simplistically speaking, with a constant distance, changing the microphone pattern affects direct to reverberant ratio.

Perhaps the easiest solution is to add ambience mikes. If you know the principles of acoustic phase cancellation, adding more mikes is theoretically a sin. However, acoustic phase cancllation does not

occur when the extra mikes are placed purely in the reverberant field, for the reverberant field is uncorrelated with the direct sound. The problem, of course, is knowing when the mikes are deep enough in the reverberant field. Proper application of the 3 to 1 rule will minimize acoustic phase cancellation. So will careful listening. The ambience mikes should be back far enough in the hall, and the hall must be sufficiently reverberant so that when these mikes are mixed into the program, no deterioration in the direct frequency response is heard, just an added warmth and increased reverberation. Sometimes halls are so dry that there is distinct, correlated sound even at the back, and ambience mikes would cause a comb filter effect.

Assuming the added ambience consists of uncorrelated reverberation, then theoretically an artificial reverberation chamber should accomplish similar results to those obtained with ambience microphones. The answer is a qualified yes, assuming the artificial reverberation chamber sounds very good and consonant with the sound of the original recording hall.

What happens to the depth and distance picture of the orchestra as the ambience is added? In general, the front-to-back depth of the orchestra remains the same or increases minimally, but the apparent overall distance increases as more reverberation is mixed in. The change in depth may not be linear for the whole orchestra since the instruments with more dominant high frequencies may seem to remain closer even with added reverberation.

The Influence of Hall Characteristics on Recorded Front-To-Back Depth

Live Halls

In general, the more reverberant the hall, the farther back the rear of the orchestra will seem, given a fixed microphone distance. In one problem hall the reverberation is much greater in the upper bass frequency region, particularly around 150 to 300 Hz.

A string quartet usually places the cello in the back. Since that instrument is very rich in the upper bass region, in this problem hall the cello always sounds farther away from the mikes than the second violin, which is located at his right. Strangely enough, a concert-goer in this hall does not notice the extra sonic distance because his strong visual sense locates the cello easily and does not allow him to notice an incongruity. When he closes his eyes, however, the astute listener notices that, yes, the cello sounds farther back than it looks!

It is therefore rather difficult to get a proper depth picture with a pair of microphones in this problem hall. Depth seems to increase almost exponentially when low frequency instruments are placed only a few feet away. It is especially difficult to record a piano quintet in this hall because the low end of the piano excites the room and seems hard to locate spatially. The problem is aggravated when the piano is on half-stick, cutting down the high frequency definition of the instrument.

The miking solution I choose for this problem is a compromise; close mike the piano, and mix this with a panning position identical to the piano's virtual image arriving from the main mike pair. I can only add a small portion of this close mike before the apparent level of the piano is taken above the balance a listener would hear in the hall. The close mike helps solidify the image and locate the piano. It gives the listener a little more direct sound on which to focus.

Very Dead Rooms

Can minimalist techniques work in a dead studio? Not very well. My observations are that simple miking has no advantage over multiple miking in a dead room. I once recorded a horn overdub in a dead room, with six tracks of close mikes and two for a more distant stereo pair. In this dead room

there were no significant differences between the sound of this "minimalist" pair, and six multiple mono close up mikes! The close mikes were, of course, carefully equalized, leveled and panned from left to right. This was a surprising discovery, and it points out the importance of good hall acoustics on a musical sound. In other words, when there are no significant early reflections, you might as well choose multiple miking, with its attendant post-production balance advantages.

Miking Techniques and the Depth Picture

The various simple miking techniques reveal depth to greater or lesser degree. Microphone patterns which have out of phase lobes (e.g., hypercardioid and figure-8) can produce an uncanny holographic quality when used in properly angled pairs. Even tightly-spaced (coincident) figure-8s can give as much of a depth picture as spaced omnis. But coincident miking reduces time ambiguity between left and right channels, and sometimes we seek that very ambiguity. Thus, there is no single ideal minimalist technique for good depth, and you should become familiar with the relative effects on depth caused by changing mike spacing, patterns, and angles. For example, with any given mike pattern, the farther apart the microphones of a pair, the wider the stereo image of the ensemble. Instruments near the sides tend to pull more left or right. Center instruments tend to get wider and more diffuse in their image picture, harder to locate or focus spatially.

The technical reasons for this are tied in to the Haas effect for delays of under approximately 5 ms. vs. significantly longer delays. With very short delays between two spatially located sources, the image location becomes ambiguous. A listener can experiment with this effect by mistuning the azimuth on an analog two-track machine and playing a mono tape over a well-focused stereo speaker system. When the azimuth is correct, the center image will be tight and defined. When the azimuth is mistuned, the center image will get wider and acoustically out of focus. Similar problems can (and do) occur with the mike-to-mike time delays always present in spaced-pair techniques.

The Front-to-back Picture with Spaced Microphones

I have found that when spaced mike pairs are used, the depth picture also appears to increase, especially in the center. For example, the front line of a chorus will no longer seem straight. Instead, it appears to be on an arc bowing away from the listener in the middle. If soloists are placed at the left and right sides of this chorus instead of in the middle, a rather pleasant and workable artificial depth effect will occur. Therefore, do not overrule the use of spaced-pair techniques. Adding a third omnidirectional mike in the center of two other omnis can stabilize the center image, and proportionally reduces center depth.

Multiple Miking Techniques

I have described how multiple close mikes destroy the depth picture; in general I stand behind that statement. But soloists do exist in orchestras, and for many reasons, they are not always positioned in front of the group. When looking for a natural depth picture, try to move the soloists closer instead of adding additional mikes, which can cause acoustic phase cancellation. But when the soloist cannot be moved, plays too softly, or when hall acoustics make him sound too far back, then a close mike or mikes (known as spot mikes) must be added. When the close solo mikes are a properly placed stereo pair and the hall is not too dead, the depth image will seem more natural than one obtained with a single solo mike.

Apply the 3 to 1 rule. Also, listen closely for frequency response problems when the close mike is mixed in. As noted, the live hall is more forgiving. The close mike (not surprisingly) will appear to bring the solo instrument closer to the listener. If this practice is not overdone, the effect is not a problem as long as musical balance is maintained, and the close mike levels are not changed during

the performance. We've all heard recordings made with this disconcerting practice. Trumpets on roller skates?

Delay Mixing

At first thought, adding a delay to the close mike seems attractive. While this delay will synchronize the direct sound of that instrument with the direct sound of that instrument arriving at the front mikes, the single delay line cannot effectively simulate the other delays of the multiple early room reflections surrounding the soloist. The multiple early reflections arrive at the distant mikes and contribute to direction and depth. They do not arrive at the close mike with significant amplitude compared to the direct sound entering the close mike. Therefore, while delay mixing may help, it is not a panacea.

Influence Of The Control Room Environment On Perceived Depth

At this point, many engineers may say, "I've never noticed depth in my control room!" The widespread practice of placing near-field monitors on the meter bridges of consoles kills almost all sense of depth. Comb-filtering and sympathetic vibrations from nearby surfaces destroy the perception of delicate time and spatial cues. The recent advent of smaller virtual control surfaces has helped reduce the size of consoles, but seek advice from an expert acoustician if you want to appreciate and manipulate depth in your recordings. We should all do this before we expand to multi-channel, for we still have a lot to learn about taking advantage of the hidden depth in 2-channel recordings.

Examples To Check Out

Standard multitrack music recording techniques make it difficult for engineers to achieve depth in their recordings. Mixdown tricks with reverb and delay may help, but good engineers realize that the best trick is no trick: Record as much as you can by using stereo pairs in a live room. Here are some examples of audiophile records I've recorded that purposely take advantage of depth and space, both foreground and background, on Chesky Records. Sara K. Hobo, Chesky JD155. Check out the percussion on track 3, *Brick House*.. Johnny Frigo, Debut of a Legend, Chesky JD119. Check out the sound of the drums and the sax on track 9, *I Love Paris.* Ana Caram, The Other Side of Jobim, Chesky JD73. Check out the percussion, cello and sax on *Correnteza*. Carlos Heredia, Gypsy Flamenco, Chesky WO126. Play it loud! And listen to track 1 for the sound of the background singers and handclaps. Phil Woods, Astor and Elis, Chesky JD146, for the natural-sounding combination of intimacy and depth of the jazz ensemble.

Technological Impediments to Capturing Recorded Depth

Depth is the first thing to suffer when low-resolution technology is used. Here is a list of some of the technical practices that when misused, or accumulated, can contribute to a boringly flat, depthless recorded picture: Multitrack and multimike techniques, small/dead recording studios, low resolution recording media , amplitude compression , improper use of dithering, cumulative digital processing, and low-resolution digital processing (e.g., using single-precision as opposed to double or higher-precision equalizers). When recording, mixing and mastering-use the best miking techniques, room acoustics, and highest resolution technology, and you'll resurrect the missing depth in your recordings.

Return to HOME DIGITAL DOMAIN

Credits:

Thanks to:

My assistant, David Holzmann, for transcribing my original 1981 article, which I have herein revised and updated for the 1990's.

Lou Burroughs, whose 1974 book Microphones: Design and Application, now out of print, is still one of the prime references on this subject and covers the topic of acoustic phase cancellation. Burroughs invented the 3-to-1 rule, expressed simply: *When a sound source is picked up by one microphone and also "leaking" into another microphone that is mixed to the same channel, make sure the second microphone is at least 3 times the distance from the sound source as the first.*

E. Roerback Madsen, whose article "Extraction of Ambiance Information from Ordinary Recordings" can be found in the 1970 October issue of the Journal of the Audio Engineering Society. Covers the Haas effect and its correlary.

Don Davis, who first defined "critical distance" and many other acoustic terms.

The Roland VM-C7200 digital mixer console with the optional MB-24 level meter bridge.

WELCOME TO THE DIGITAL AGE

This is the digital age. The advantages of creating, editing and producing music in the digital domain are very well known. Virtual tracks, non destructive editing, and powerful on board digital FX have made Roland's VS series of Digital Studio Workstations the most successful digital studio workstations in history!

But what about mixing? What is digital mixing? Does it have the same kind of advantages that digital recording and editing have over the old analog ways? And how does a digital mixer work, anyway. Isn't mixing just moving a fader up and down? So why would a digital mixer be any better than an analog mixer?

Let's answer that question and take a look at what incredible advantages digital mixers can offer you and your music. Along the way, we'll discuss some of the basic features and concepts of mixers in general to help show how the flexibility and power of digital mixers is truly revolutionary.

Digital Mixer VS Analog Mixer Overview

Before we get started looking at mixers in general, here are a few of the major benefits digital mixers have over analog mixing.

FLEXIBLE CONFIGURATION VERSUS FIXED PATHS

In an analog mixer, all of the connections, inputs and outputs, are "hardwired". Once the design is finished, the mixer can never change.

In a digital mixer, once the audio is inside the mixer, there is virtually total freedom to move it around, add effects, and configure its paths anyway you need to for your application. For example, any input can go to any or all channels. And the various paths within a digital mixer can be routed to many different destinations as well as different physical outputs.

SIZE AND PRICE

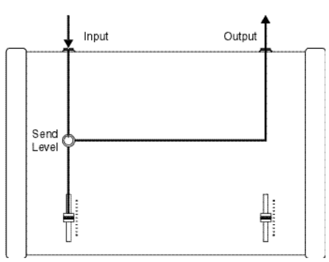
Because the configuration and control of a digital mixer is so flexible, digital mixers can be designed in more compact forms to accommodate the size and shape you need for your studio. For example, an analog console with 94 channels of audio would be very, very wide, probably over 12 or more feet, and would weight thousands of pounds.

Because of the flexible control options a digital mixer offers, you could control 94 channels using a much more convenient sized console that could still be portable.

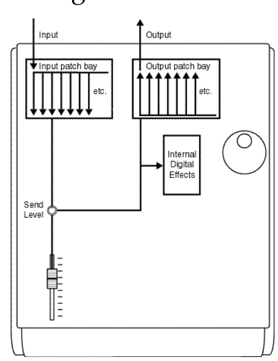
This design flexibility also allows digital mixers to often be a lot less expensive than an analog console with the same number of channels.

POWERFUL ONBOARD DIGITAL EFFECTS

Because the audio in a digital console has entered the digital domain, it is easy and cost effective to add very high quality, on-board digital effects processing. Since this processing is an integral part of the mixer, you avoid the audio losses of cabling, external patching and audio conversions inherent in an analog console.

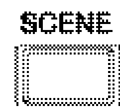


Analog Hardwired Audio Paths vs Digital Flexibility



TOTAL RECALL OF ALL SETTINGS

A huge advantage of digital consoles is their ability to store and then recall all of the settings of a mix, even including the effects. This allows you to work with a project, store its settings and later come back to the mix exactly as you left it.



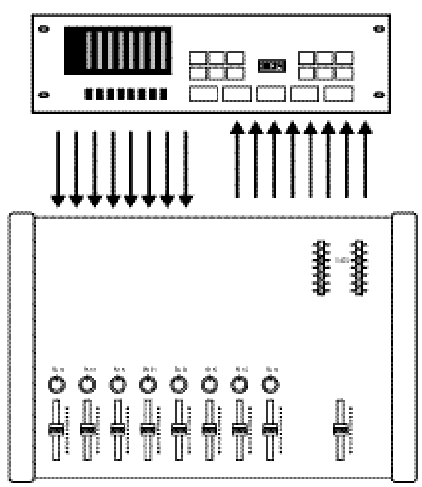
Easy recall of all mixer settings

A mixer with memory has lots of other advantages. For example, you could recall a vocal's EQ and level for a new take or to punch in over an old version with matching audio quality. The speed of recall also allows you to try one mixing approach, store it, try a completely different idea, and then compare the two. This really helps your music sound better.

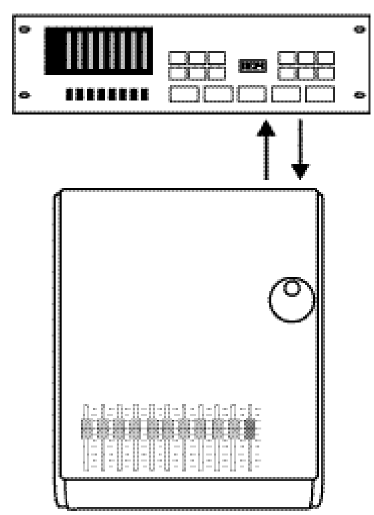
DIRECT DIGITAL CONNECTIONS TO TAPE RECORDERS OR DATS

In today's music world, more and more of the multi track and stereo recorders are digital. When you are processing and mixing your music in a digital mixer, you have direct digital paths to both digital recorders and digital mix down devices such as DATs. This avoids the losses inherent in A/D and D/A conversions that would be required using an analog mixer. Again, your music will sound better.

But let's look at some more basic mixer concepts to help us see what some of the advantages of digital mixing are all about.



Many Analog Connections

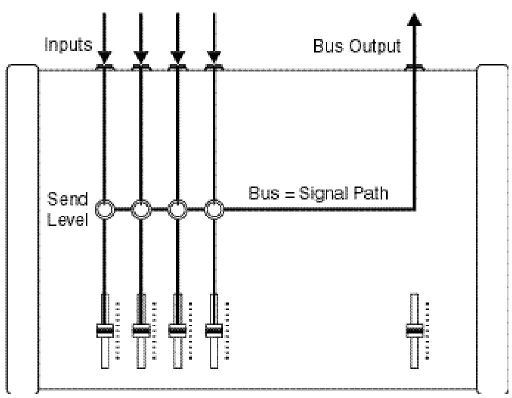


Two Digital Connections

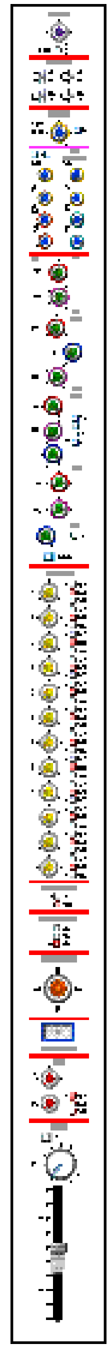
Definition of a Mixer and Busses

Before we go any further, we really should define a mixer and its most basic parts and functions. Actually, a mixer is a very simple device. It takes audio from various inputs and sends this audio to various outputs. Along the way it can combine the audio streams, change their levels, and sometimes even change the audio using processing.
The basic controls for each stream of audio are called CHANNELS. A channel contains the controls to send a stream of audio along various paths and to change the level and process the audio.

The paths that audio is routed along in a mixer are called BUSSES. A bus is a path that can be accessed by more than one stream of audio and that goes to one or more destinations. Sometimes two busses are combined and used as a stereo bus routed to two outputs for stereo applications.



Simple Signal Bus



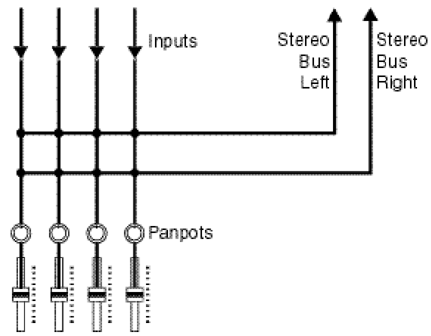
Digital Channel Strip

Types of Busses and Their Applications

In traditional analog mixers, every path is hardwired, as we discussed earlier. There are several different ways that busses have traditionally been wired.

STEREO MASTER BUS

This bus is used as the main outputs for a mixer and often is sent to a two track tape recorder or the analog inputs of a DAT. The main fader for a channel is used to adjust the send levels to this master bus.

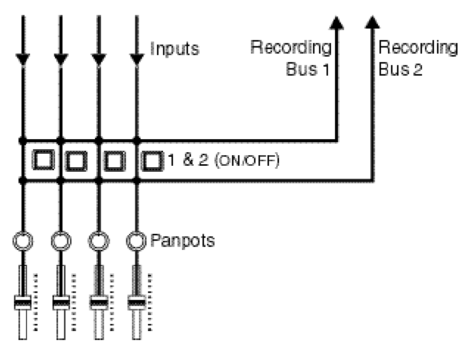


CUE AND MONITOR BUSSES

These can also be stereo busses and are used to send headphone or monitor mixes to performers or audio to the speakers and headphones in a control room.

GROUP, SUB GROUP OR RECORDING BUSSES

These busses are traditionally used to send audio to tape recorders. They are arranged in pairs and are accessed using an On/Off button. There is no send level so the volume of the channel sent to the bus just follows the level of the main channel fader. The balance of audio sent to either pair is adjusted using the Channel's PAN control.
Because there is no independent level control for these busses, their use is limited.



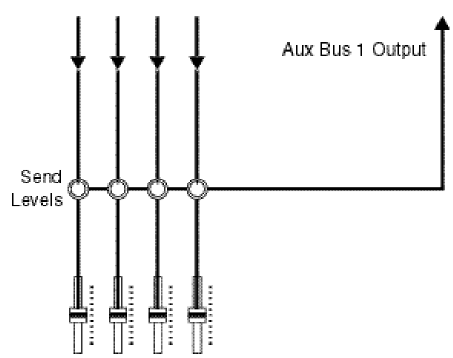
Types of Busses and Their Applications

AUX BUSSES

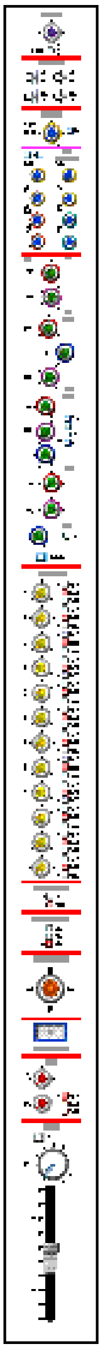
Aux busses are mono or sometimes stereo busses that are accessed using a small level control knob. Aux busses are routed to a dedicated output and are used for either headphone mixes or for sending audio to external FX processing.

Aux busses are more flexible than Recording busses, as they have send levels. They sometimes also have a switch that allows their level to be independent of the main channel level. This switch sends the channel's audio to the Aux bus either PRE: independent of the main fader; or POST main fader.

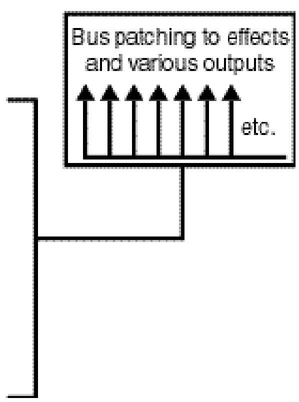
With the PRE fader position, the mix out of the Aux bus can be used for headphones or monitor sends since the output levels for each channel will be independent of the main or stereo output level determined by the channel's main fader. In the POST fader position, the send level will follow the main master bus levels, which is often used for sending audio to external FX processors.



A Digital Mixer’s More Flexible Busses



In an analog console, all of the channels and busses have fixed, hardwired paths. For this reason, some analog consoles are designed for mixing and recording purposes and others for live mixing. Most of the design difference is based on the busses and how they will be used.
Some powerful digital mixers have busses that are much more flexible than analog busses. In fact, these are sometimes called Flex Busses! These flexible busses combine all of the features of every type of analog bus. They have send levels and can be sent PRE or POST fader (and even from more locations along the channel strip). The outputs of a Flex Bus can also be sent to many different destinations as well.



For this reason, a console with a very flexible bus structure can be configured by the user for just about any mixer application imaginable. They can be used for sending audio to internal or external effects, to either analog or digital recorders, to headphone amplifiers, to monitors or studio speakers and can even be sent to other busses to be used as "matrix" busses for live sound.

This kind of configurable bus structure is one of the great examples of how digital mixers can be more "flexible" and powerful than any type of analog mixer.

Inputs and Outputs

A mixer combines and routes audio that comes from inputs and then sends this audio to outputs. Let's take a look at the types of inputs and outputs available to most mixers.

INPUTS AND SOURCES

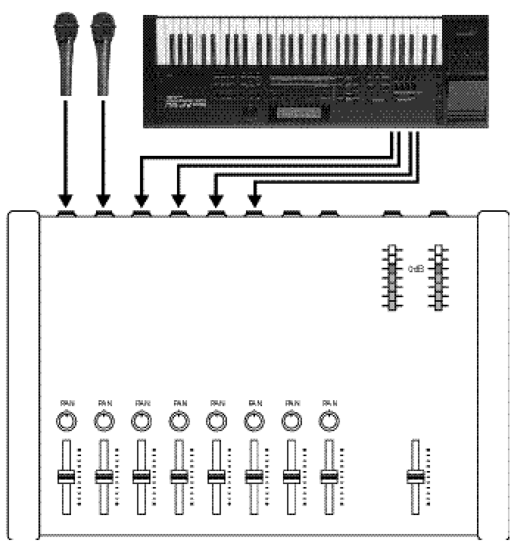
Low level sources require low level mixer inputs. Typical low level inputs include:

- Microphones are a standard in the studio. Most have XLR balanced connectors and require mic level inputs. Some require extra (phantom) power directly from the mixer via the mic cable.
- Guitars require low level mixer capability also.

High level sources require higher level mixer inputs, such as:

- Synths and electronic instruments
- Tape recorders
- Effects processor outputs

In today's world, most studios need lots of channels of both mic and line level inputs. For live applications, especially, you need lots of mic inputs.



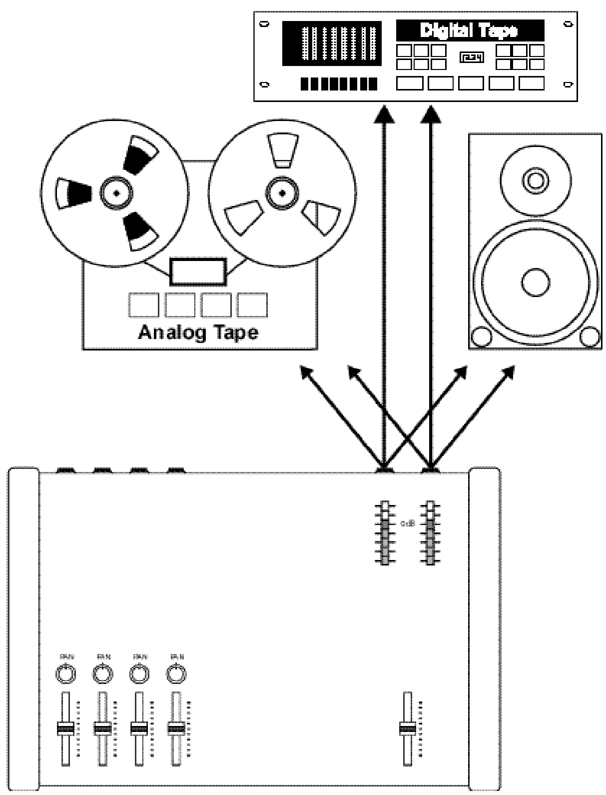
Outputs and Destinations

Audio from a mixer can go to many different destinations, depending on the application. Almost always these outputs are high or line level.

Typical output destinations would include:

- Mix down tape recorder via Master Bus outputs
- Multi track tape recorders via Recording Bus outputs
- External effects processor via Aux outputs
- Surround sound amplifiers or encoders via various Bus outputs
- Headphone amplifiers via Aux outputs
- Stage Monitors
- Studio Monitors
- Various speakers and their amplifiers

These outputs are usually RCA, 1/4 inch phone, or XLR connectors.



Inputs and Outputs: The Digital Difference

In an analog mixer, all of the paths from the inputs to the channels to the outputs are "hardwired" and can't be changed. An aux bus only goes to the aux send output. Period.

So what if you need extra Pre Fader headphone sends, need to do a Surround Sound mix, or need to send lots of extra channels to a multi track recorder?

In most digital mixers, the routing is totally flexible. This includes the paths from the inputs to the channels, the paths from the channels to the busses, and via the busses to all of the outputs.

For example, in some digital mixers, any physical input can go to any or even all tracks. This way you can route a vocal to one channel to process without EQ or compressors and to another channel with some EQ, compression and delay for an easy way to record and compare two differently processed "takes".

In digital mixers with a flexible bus structure, you can route your audio anyway you want to your busses, depending on your needs at the time.

The outputs of most digital mixers are also completely flexible. For example, a bus can be used for a headphone mix and sent to a monitor output for one application, and then later used as a recording bus and sent to a set of assignable outputs or even sent digitally to an MDM (Modular Digital Multitrack). The bus could also be used to send audio to an external FX processor or even for Surround Sound mixing via balanced outputs.

The routing on a digital mixer is not fixed. It's available for use for whatever application you need.

Connecting to the Digital World the Digital Way

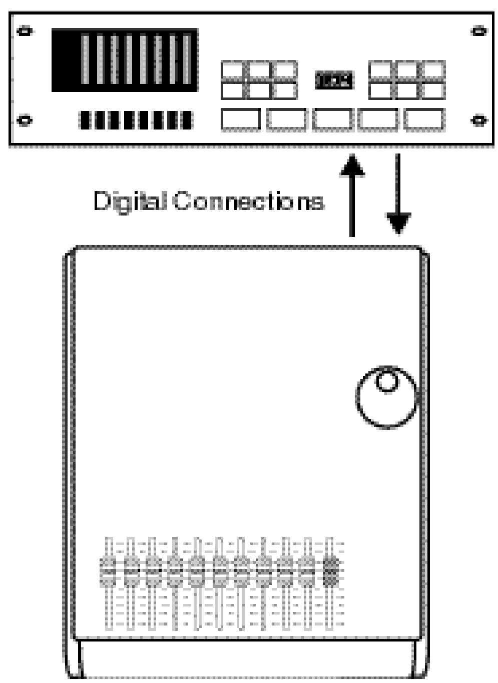
Of course in an analog console, all of the inputs and outputs are analog. This makes it difficult to connect an analog console to various digital devices such as digital effects units, digital tape recorders, or digital mix down decks such as DATs.
Digital consoles have, in addition to many analog inputs and outputs, many different digital connectors including stereo digital bus outputs for routing to digital FX or DATs. In today's studio, more and more mixing is being done digitally to DAT. This, of course, can be done even at a 24 bit level from many digital mixers.

Digital routing to MDMs is also possible from most digital mixers. This gives you high quality audio and total flexibility with out the hassles and audio losses of lots of cabling, patching and useless A/D and D/A conversions.

Routing to digital devices from digital mixers is easy, convenient, and keeps your audio at its highest quality.

Avoiding Conversions

Keeping your audio in the digital domain avoids the audio losses from numerous A/D and D/A conversions and ensures the highest quality audio for your music.



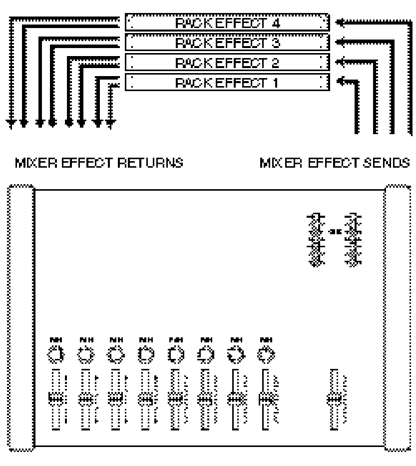
Internal Digital Effects

A huge part of mixing audio for any application is adding FX. In a digital mixer, these effects can be all digital, right inside the mixer. This gives you much higher audio quality, without the patching hassles of analog FX and analog patchbays.

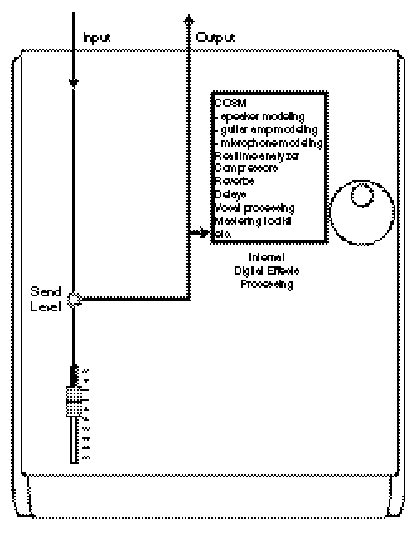
The internal processing power in a digital mixer can also be reconfigured. For example, you could have a Real Time Analyzer for live concert work or when you are first tweaking the sound in your studio. When you’re recording guitars you could have COSM guitar amp and speaker models to quickly and easily get the guitar sound you need without using lots of loud studio gear bleeding into other mics or just driving your neighbors crazy. For live gigs, you could configure your console so you have compressors on 20 or 30 inputs just to make mixing the gig a bit easier.

When you’re mixing a music project, you could use COSM Speaker Modeling to listen to your mix as it would sound on a variety of different kinds of speakers. Later you could configure your effects to be a Mastering Tool Kit with multi-band compressors and expanders as you mix digitally to CD or DAT.

All of this is possible in a digital mixer. The convenience and power of onboard digital effects really can enhance the quality of your music.



External Effects



Internal Effects

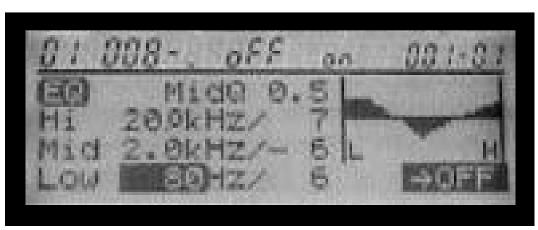
Instant Information and Fast Access

The audio in a digital mixer is in digital form. That makes it easy to display the audio in many convenient ways that are not possible with analog mixers. For example, you could see the interaction of all of the EQs on one channel as a graphic of the whole EQ. This gives you instant feedback on exactly how you are processing your channel. You could also call up a screen or even a bank of moving faders that instantly shows all of the send levels to one of your busses. This is much easier than looking down a row of tiny knobs and trying to see what their settings are.

And controlling your audio can be much faster on a digital console. For example, suppose you have a 94 channel analog system and want to change the send level to a certain bus for channel 2. To do this, you actually have to get up out of your chair (remember, this console is going to be over 12 or 13 feet long!), walk down and count the knobs to find the bus you want to change. All this time you have left the “sweet spot” of your speakers, so you can’t even really hear the change you are making while you are making it.

On a digital console, you could access the send to that bus by pressing one button. All of your parameters for that channel or that bus would be right in front of you for fast and easy access. AND, you would still be comfortably in your seat, right in the correct listening position, so you would be able to more accurately make your changes.

Digital consoles make mixing easier!



Memory: Scenes, EZ Routing and Libraries

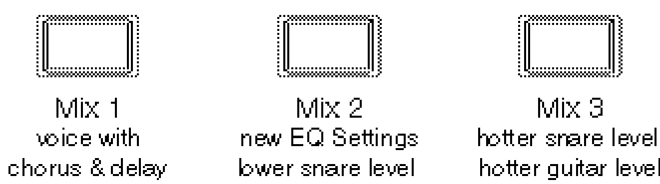
As you learn more about digital mixers, you can see how powerful and flexible they really are. To help you manage your mixing, some digital mixers have features designed especially to help you reconfigure your console for the type of work you are doing, or, to get back to an earlier project exactly as you left it. These are all advantages not available on analog mixers.

LIBRARIES

A digital mixer can have lots of powerful processing on each channel such as many bands of digital EQ or dynamics processing. To help speed up working on a project, a digital mixer may have libraries of different EQ settings or dynamics presets. These libraries are available to use in any project, so you can store your favorite EQ for recording a bass player or a drum set, for instance. You can also use the presets to help you get started processing your audio.

SCENES

Scenes are pictures of every parameter of your digital mixer: from the levels to the EQ to the FX settings to the bus routings; everything. Scenes let you recall a mix just as you left it, so you can get right to work on it again. Scenes allow you to bring back the exact settings you used when you recorded a vocal, so you can overdub with the same EQ and levels. Scenes let you store your current mix idea, work on another great idea you just had, then compare the two. They can also let you instantly reconfigure your mixer for live concerts if you need different levels, FX or mic settings between songs, or from one band to another. Scenes can really help you get better results whether you are mixing or tracking or overdubbing or doing live concerts.



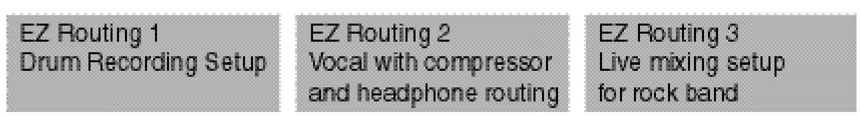
EZ ROUTING

Some digital mixers have a powerful function called EZ Routing. EZ Routing has two different basic applications. Whereas Scenes are pictures of all of your mixer settings for one particular project, EZ Routing templates are complete, global configurations of your mixer that can be called up at any time in any project.

For example, you can store templates of how you like to record drums, with the exact FX, EQ, and bussing. You can use EZ Routing templates to instantly re-configure your mixer from a recording mixer to a mastering mixer to a live sound mixer. All with the push of one button.

EZ Routing also has another valuable function. It can take you step by step through all of the different settings you need to set up your mixer for just about any application. In this capacity, it’s like having an onboard guide or help system. Mixing has never been easier and more flexible.

And of course, this power to instantly recall or reconfigure your console is not available on an analog console.

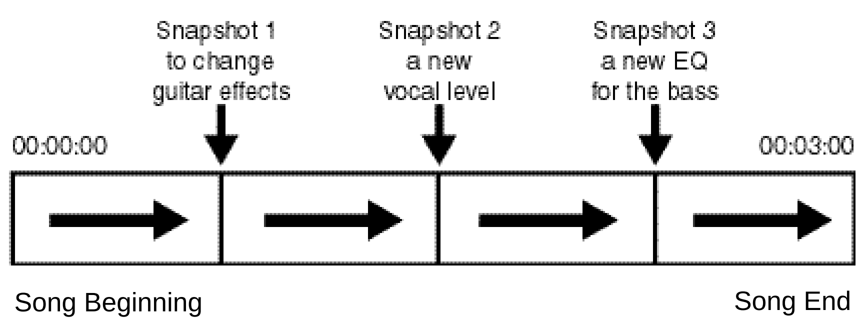


Use EZ Routing to store custom routing templates

Automation: Dynamic, Snapshots & Moving Faders

Picture yourself doing a complicated mix. You've got the vocal level changes down pretty well, but how are you ever going to be able to keep track of those changes and still work on the drums? Once again, digital mixers come to the rescue with their automation functions. Like the Scenes described above, a mixer's automation memory can recall the various settings of your mixer. Unlike scenes, however, which can be recalled manually at any time by the mixing engineer, automation changes happens automatically at a certain time during the playback of your song.

There are two basic types of automation: snapshot and dynamic. Snapshots are also complete pictures of your mixer: all of the effects settings, levels, etc. Some digital mixers let you customize which mixer parameters are stored in your snapshots. You place a snapshot where you need an effects patch change on your guitar part, an instant volume change for verse to chorus vocal or anywhere you want an immediate mixer adjustment. When the song plays back: voila: the change happens automatically. Snapshots are also great for using the same internal FX processor in your digital mixer for different effects at different times.

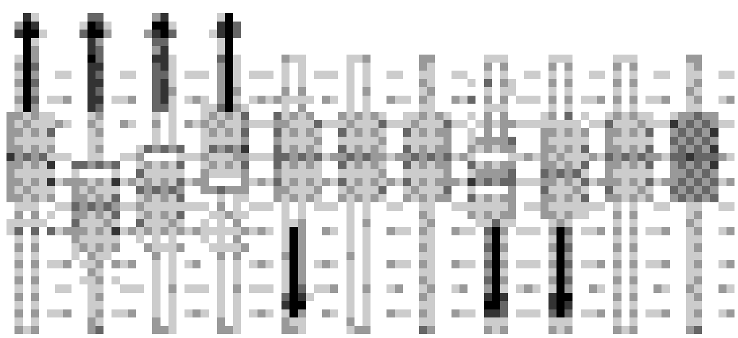


Automation: Dynamic, Snapshots & Moving Faders

Dynamic automation is used when you need gradual changes of parameters such as a fade out. For example, you could use dynamic automation to recall every little level move you make to your vocal track. Once you have it right, you can then move on and work on the drums, knowing that the vocal track is now perfect.
Some digital consoles even have moving faders that help you see exactly what is happening during your mix.
The memory for this automation in your digital mixer can be on board or it can be transmitted via MIDI. If it is MIDI based automation, then you would use a sequencer to play back the automation data during your mix.

It's easy to see how automation could really help you perfect your mixes, working on just one part at a time and getting everything exactly the way you want it.

Automation is another great way that digital mixers enhance the sound of your projects.

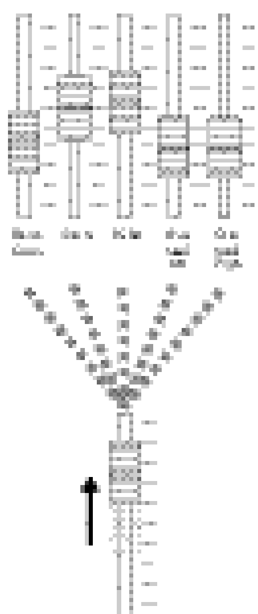


Channel Grouping and Linking

Digital mixers offer some new, great ways to control your audio that will help you make better mixes. One way is Channel Linking. If you have two channels that are being used together as a stereo pair, you can link the volume controls together so you can control them using only one fader. This helps with the accuracy of your mix. You can often use this channel linking and still maintain independent pan positions or other channel parameters.

Fader grouping is an extension of linking. Using grouping, you can control the level of several different channels using only one fader. This is a great way to control the mix of a whole drum set or several background vocals by using only one fader. This is much more convenient and accurate than trying to bump up all 8 drum mics a little bit using 8 faders (unless you are related to an octopus!). Of course with fader grouping, you can set the relative balance of each part before you group them. That way your overheads will all stay in balance with your snare part, but you can raise or lower the level of your whole drum kit using only one fader.

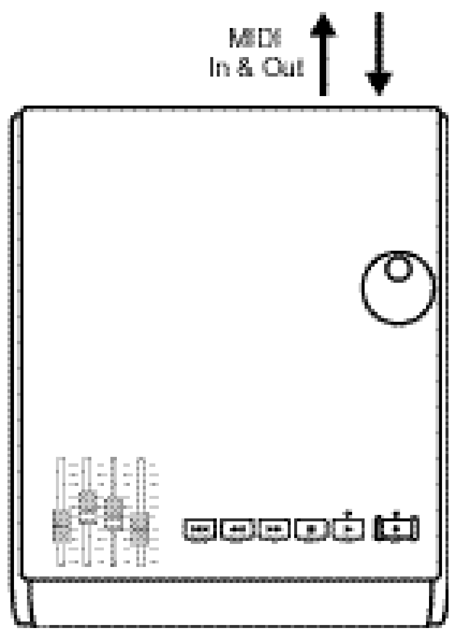
Digital mixing definitely makes mixing easier!



One fader can control a whole drum mix

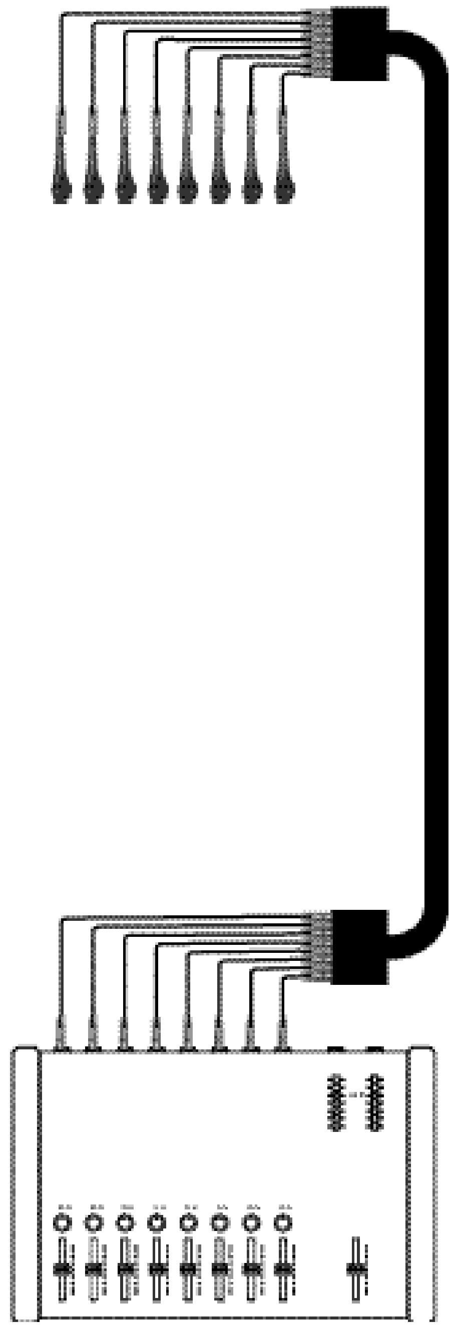
Since the functions of a digital console are really limited only by the imagination of the software engineers, there are lots of new and powerful applications for a digital console. One great application is the control of MIDI devices right from the console. This allows you to control the levels of audio in computer based systems that might not have digital audio I/O. You can also control the transport functions of sequencers and MDMs, arm tracks, and have a completely integrated control surface right from your mixer.

Controlling the volume of computer based audio or sequenced parts right from your mixer is especially powerful, as mixing with a mouse is kind of like trying to tie a knot using only one finger!!



Sending MIDI information from mixer faders and transport controls

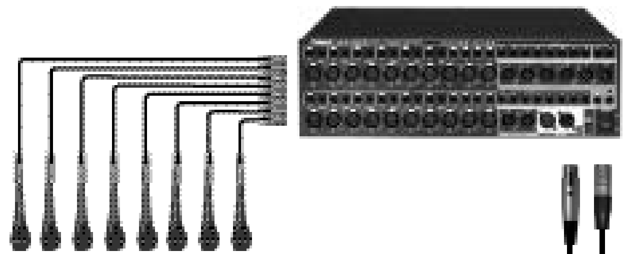
Console Designs

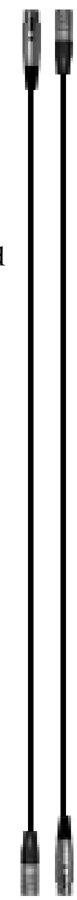


Analog consoles are hardwired as mentioned above. The inputs are hardwired to the channels which are hardwired to the busses which are hardwired to their outputs. For this reason, the physical connections to an analog mixer are made right on the mixer itself. Usually these are on the rear of the mixer, but sometimes they are made right on the upper surface of the console.

Of course the sources for the mixer such as microphones, tape recorders, effects processor are usually located quite some distance from the mixer. This requires that long analog snakes be made up to bring all of the audio right into the control room to connect into the mixer. As we know, analog cabling, especially long runs, can introduce hums, buzzes and other types of signal degradation.

This type of wiring is also extremely time consuming and can be very expensive, especially when it comes time to try and eliminate the worst of those hums! In addition, once the console has been wired in place, there are so many wires and cables around that the console is impossible to move.





With digital consoles, all of the audio is controlled in a flexible system that is not hard wired at all.. For this reason, some advanced digital mixers actually have a control surface that is totally separate from the inputs and outputs of the mixer. This means the I/O interface, usually called a processor, can be located right where your audio sources are located. If your mics are in a studio, then the processor can be there. If you are doing a live concert, then the processor can be right on stage. The only connection you need back to your mixing control surface or console are 2 digital cables. NO SNAKES!!!!

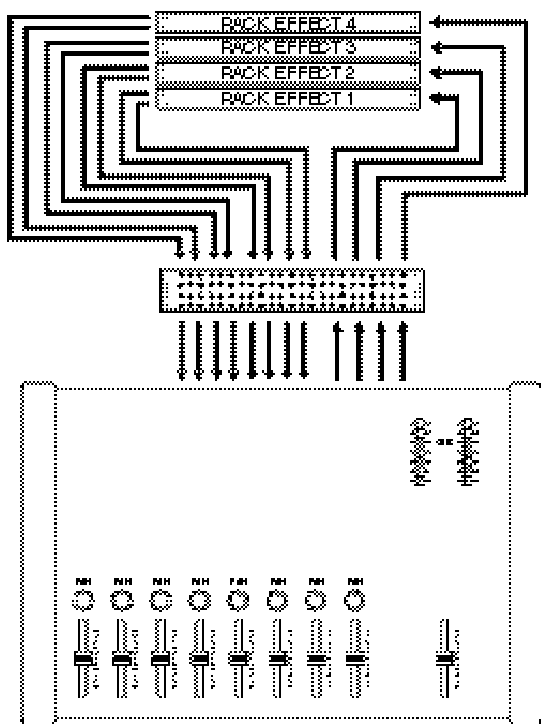
Of course you can see the incredible advantages this offers the mixing engineer or musician. First of all, you don't have any signal losses from the stage or studio to the mixer. All of the cable runs are as short as possible, eliminating most of the wiring problems that studios typically have. Secondly, now that there aren't mountains of cables and lines and snakes ending up in your control room, you can actually have the freedom to move the console around to a new configuration, or just move it out of the way if you have some friends over for a jam session. Your studio now becomes custom configurable, just like your digital mixer! Finally, setting up your studio or your live mixing system is much easier, faster and less expensive.

Eliminating analog snakes is a big advantage of some digital mixers.

No External Patchbays and Wires

Of course you have seen the patchbays in most studios with analog consoles. These are used to help the engineer route their various inputs to different channels, to patch in effects as needed, to patch tape decks to the different outputs of the mixer and so on. And you have probably seen the confusing rats nest of cabling that goes with such a patchbay. Of course it goes without saying that having lots of extra audio connections with their capability of generating extra noise isn't exactly what the doctor ordered for your mixes.

Well digital consoles can help you eliminate these patchbays, and their expense and wiring hassles. Since digital consoles have built in, digital patchbays, you can route different microphones to different channels, add effects, patch busses to tape recorders digitally, all without using a single analog cable. This is a very elegant and clean way to work with your audio!



Unlike this analog mixer with an external patchbay and lots of wires, digital mixing with internal digital effects requires no extra wiring!

Other Cool Digital Mixer Tricks

Finally, since digital mixers are controlling digital audio, the sky's the limit when it comes to processing your audio. In fact, instead of having an analog console that is basically getting older by the minute, digital consoles can have software updates, adding features and making them "younger as they get older"!!!

How about adding a real time analyzer, or controlling the muting of several faders for your drum kit using 1 mute group button? Accessing commonly used functions using one macro button? All of this is easy using a digital console. Special effects such as powerful 24 bit mastering tools, guitar amp and speaker models, microphone models and even speaker modeling are possible once your audio is in the digital environment and surrounded by powerful effects processors. And of course new effects patches can be added easily. How about additional MIDI functions for controlling external FX processors in real time, allowing internal or external real time FX editing? Doing live sound, how about a built in feedback eliminator?

The list will go on and one as more features are dreamed up by designers or even users. The bottom line is that features and powerful processing can be added via software to help your digital mixer become even more and more powerful.

Summary

Digital mixers offer extensive advantages over analog mixing. Once you see how these new powerful features can help you easily create better sounding and more accurate mixes, you'll never want to go back to the old analog days.

Digital mixer features that will improve your mixing include:

- **Configurable input and output routing**: any channel can go anywhere
- **Flexible bus structure** for the maximum mixing power
- **Direct digital connections** to DATs or MDMs
- **Lots of channels** in **compact designs**
- **Easy and fast** operation.
- **Powerful onboard digital effects** for high quality and repeatable FX processing
- **Internal digital patchbays** for easy, high quality routing
- **Total recall of all mixer settings** using Scenes
- **Instant configuration of mixer FX and routing** for different applications
- **EZ routing** for on-screen help and storing your commonly used mixer settings
- **Scenes, snapshots and dynamic automation** to help you perfect your mixes
- **MIDI fader and transport buttons** for easy control of MIDI sequencers and workstations
- New and powerful effects built in such as **Real Time Analyzers and Speaker Modeling**
- **Channel grouping, linking, mute groups** and other powerful ways to help you mix
- **Separate console and processor** to eliminate the audio losses of analog snakes

The list goes on and on, but the bottom line is that digital mixing can give you the tools to ensure that your audio is the highest quality possible and to help you make better mixes.

Roland VM-3100 and VM-3100Pro

Inexpensive, compact with 24 bit audio quality, the VM-3100 has made digital mixing affordable enough so anyone can have the advantages of digital mixing.

- Affordable 12/20* channel 8 bus digital mixer
- 24-bit AD / DA
- 3 band 24 bit digital EQ per channel
- 1 or 2* onboard stereo effects processor with compression, reverb, chorus, delay, and guitar/keyboard/vocal multi-effects, COSM Speaker and Microphone Modeling
- 2 additional mono compressors
- Digital I/O
- Scene and EQ memory
- EZ Routing
- 8 Bus outputs
- 16 MIDI fader function and MIDI transport controls
- Full MIDI automation using external sequencer, including switching scenes, mixer and fader levels
- Optional DIF-AT Interface Box for 8 channels of digital I/O for ADAT/Tascam MDMs*

*applies to VM-3100Pro only

Roland VM-7000 series 94 channel digital mixers**

The VM-7000 series offers more power than ever before available for any kind of mixer, either analog of digital at anywhere near the price!

- 94 Channels of digital mixing
- 40 mic / line inputs
- 24 Bit A/Ds
- 32 mono or 16 stereo digital FX processors on board
- 94 DSP blocks for use as extra EQ or dynamics processing
- Separate console and processor eliminating noisy analog snakes
- Modular system with choices in processor and control surfaces
- Up to 48 tracks of digital routing to MDMs for recording and mixing
- On board dynamic automation using moving faders
- Scenes, libraries, EZ Routing
- Flex Buss and virtual patch bay for customizing use in many applications
- COSM speaker modeling, microphone modeling, guitar speaker and pre-amp modeling
- 5.1 Surround mixing

**specs are for fully expanded system including 6 optional VS8F-2 processing cards

FeatureComparison

	VM-3100	VM-3100 Pro	VM-7200 Series	Analog Mixers
Configurable input routing	YES	YES	YES	No
Flexible bus structure	YES	YES	YES	No
24 bit digital I/O to DAT, etc.	YES	YES	YES	No
Direct digital path to MDMs	No	YES	YES	No
Onboard digital effects	YES	YES	YES	No
COSM Speaker Modeling	No	YES	YES	No
COSM Microphone Modeling	No	YES	YES	No
Internal digital patchbay	YES	YES	YES	No
Total recall of all mixer settings	YES	YES	YES	No
Snapshot automation	Via MIDI	Via MIDI	YES	No
Dynamic automation	Via MIDI	Via MIDI	YES	$$$$
Moving Faders	No	No	YES	$$$$
Separate console and processor	No	No	YES	No
EZ Routing for configuration help	YES	YES	YES	No
MIDI Fader Function	YES	YES	YES	No
Fader linking	YES	YES	YES	No
Fader Grouping	No	No	YES	$$$$
Mute Groups	No	No	YES	$$$$
Real Time Analyzer	No	No	YES	No

A/D: For "Analog-to-Digital Converter," a device that receives analog audio and converts it into digital data, such as analog audio coming into a digital mixer

Automation: The memorization and playback of changes you make to mixer settings

Aux: Short for "Auxiliary"; a designation for extra busses typically used for sending signal to effects, headphone amps and other destinations

Aux return: An extra input; typically used for receiving a signal from the output of an internal or external effect processor

Aux send: An extra bus that can be used for sending signal anywhere; typically used for sending signal into an effect

Balanced: A type of audio connection that uses the three leads in a cable, connector and jack as part of a phase-cancellation scheme to boost signal and reduce noise

Band: In EQ, a range of frequencies

Bandwidth: In EQ, the width of a band; the number of frequencies that will be boosted or cut above and below a selected center frequency

Bus: A pathway down which one or more signals can travel

Cannon connector: Another name for an XLR connector

Channel: A set of tools for the control and shaping of a single signal

Channel strip: A row of controls on a mixer designated for the shaping of a single signal

Compressor: A dynamics processor that reduces the level of any signal exceeding a specified threshold volume

Condenser Microphone: A type of high-quality mic that requires power

COSM: An abbreviation for Roland's "Composite Object Sound Modeling" technology that shapes audio by applying the sonic characteristics of popular or classic microphones, guitars, guitar amplifiers and studio reference speakers

Cue Bus: A bus—sometimes a stereo pair of busses—dedicated to the providing of signal to performers so they can hear what they're doing

D/A: For "Digital-to-Analog Converter," a device that converts digital data to analog audio, such as the audio leaving a digital mixer on its way to an analog device

DAT: Abbreviation for "Digital Audio Tape"; used in reference to this type of tape as well as the recorders that use it

Delay: An effect in which a copy of a signal is played back later than the original

Effects: Any of a variety of audio processes that can be applied to a signal to modify it, including reverb, delay, flanging, phasing

Effect Loop: A two-way journey of a signal from a channel insert point to and effect and back to the insert point

Glossary

Effect return: An input that receives signal from the output of an internal or
EQ: A popular abbreviation for "equalization," the thing an equalizer does
Equalization: The process of altering the levels of frequencies that comprise a signal
Equalizer: A device that boost or cuts the volume of specific frequencies in a signal
EZ Routing: A re-usable template containing a Roland digital mixer's routings; in some cases, walks you through the creation of a setup using displayed questions
Fade In: A change in level over time increasing upward from silence
Fade Out: A change in level over time falling gradually to silence
Fader: A slider-type device that's used for the precise manipulation of levels
FlexBus: A powerful all-purpose bus available on Roland digital mixers
Flying fader: A motorized fader that automatically moves during automation playback
Frequency: Refers to the number of times per second that a sound wave's cycle repeats, with a greater frequency resulting in a higher perceived pitch; also used as shorthand for describing sound waves in a signal by their pitch
Gain: Another term for level
Grouping: A process by which multiple channel strips are joined together under a single level control
Hertz: (Hz) A unit of measurement equal to a sound wave's single cycle
High-pass filter: A filter that removes lower frequencies from a signal, allowing higher frequencies to pass through unaffected
Hum: An undesirable low-frequency tone present in a signal as a result of grounding problems or proximity to a power source
Impedance: The amount of force with which voltage leaves a connector and the amount of resistance to that force in the jack receiving it; they should be equal
In-line: Any effect accessed by interrupting a channel's signal flow, directing its signal to the effect, and returning the output of the effect to the channel at the same point from which it came (also called an "insert effect"); also, a mixer whose multi-track tape return controls are contained in its channel strips
Input: A jack that receives audio
Input level: The level of signal coming into a channel strip
Insert: A point in a signal flow at which an in-line effect can be employed
kHz: for "kiloHertz": a thousand Hertz
Level: A general term for volume or amplitude
Limiter: A compressor set to a ratio of 10:1 or greater

Line level: The high-level signal produced at the outputs of audio equipment such as mixers, recorders and playback devices

Macro: A digital mixer shortcut that performs a multi-step operation as a single action

Meter: A visual device that shows the level of a signal

Meter bridge: A separate piece of mixer hardware that provides an additional array of meters

Mic: A common nickname for "microphone"

Mic Level: The low-level signal produced by microphones and electric instruments such as electric guitar or bass

Microphone: A device that converts sound waves into audio signals

MIDI: For "Musical Instrument Digital Interface," the wiring and message protocol that allows musical instruments and other devices to communicate

Mixdown: Acommon synonym for the noun "mix"

Mix: As a noun, a signal that contains one or more other signals—typically a mix is a pair of stereo signals that contains numerous mono and stereo signals, along with effects, combined together; as a verb, the act of creating such a combined signal, or of using a mixer in general

Mixer: A device in which audio signals can be manipulated, enhanced and directed to other destinations, singly or together; also, someone who works a mixer

Modeling: A process by which the characteristics of one signal are applied to another; Roland's advanced COSM modeling creates realistic emulations of popular and classic microphones, guitars, guitar amplifiers and studio reference speakers

Monitor: As a noun, a speaker, or set of speakers, for the purpose of listening to a mix; as a verb, the act of listening when mixing

Mono: A single signal

MMC: For "MIDI Machine Control," the MIDI-based protocol that allows the controls of one MMC-compliant device to affect the transport mechanism of another

MTC: For "MIDI Time Code," a form of SMPTE used for the timing synchronization of two or more MIDI-compliant devices

Mute: A switch that allows you to silence a channel's signal; some mixers provide mute grouping for silencing multiple channels at once

Out of phase: A situation in which the sound-wave cycles in one signal reach their greatest amount of air pressure as the cycles in a similar signal reach their least; the two signals will cancel each other out

Outboard: External, as in an "external device"

Output: A jack that sends out signal

Overload: What occurs when a signal is so loud that it exceeds the capabilities of the device through which it's passing

Pad: A device that lowers the level of a signal

Parameter: A setting whose value can be changed

Parametric: A type of EQ that can be adjusted so that it can boost or cut any frequency within its overall range; may also have a user-definable bandwidth

Patch: A temporary connection made between two audio devices, or within one

Panning: The left/right positioning of a signal within a stereo image

Phantom Power: The power required for the operation of a condenser microphone when it's not supplied by internal batteries or a separate power supply

Phone Connector: A 1/4" connector used for the transmission of mic or line-level audio

Phono Connector: A small audio connector used for the connection of line-level signals and S/PDIF Aformat digital audio connections

Post: The designation for accessing audio just after it leaves a particular channel component for example, "post-fader" grabs audio just after it leaves the channel's main level control, before it gets to the panning control

Pre: The designation for the accessing of audio before it gets to a particular module; for example, "pre-EQ" grabs audio before it gets to a channel strip's EQ

RCA Connector: Another name for a phono connector

Reverb: An effect in which the ambience of a physical space is simulated; a signal is copied many times, and the copies are heard one after another at decreasing levels, so closely together that they are not heard individually

Return: A bus or input jack that receives signal, commonly used for effect outputs

Scene: All of a song's or project's mixer settings saved in a Roland digital mixer's memory; can be quickly recalled, re-establishing all settings instantly

Send: A bus or output jack that transmits signal

Sequencer: A MIDI recorder that captures MIDI data and can play it back in real time

Shelving: A type of EQ in which all frequencies above or below a selected frequency are affected; low shelving affects all frequencies below the selected frequency; high shelving all those above it

Signal Flow: The journey a signal takes from one place to another

Snapshot: A captured group of Roland digital mixer settings that reflect the state of the mixer at a particular moment within a song or project; mixer can recall the snapshot and re-instate its settings at the proper moment during subsequent song or project playbacks

Solo: When monitoring, the isolation of one signal by silencing all other signals
Synchronization: Or "sync"; the coordination of timing between audio and/or video devices
Take: An attempt at recording something; each try is called a "take"
Track: A stream of recorded audio data
Treble: The higher frequencies in a signal
Unbalanced: A type of connection that utilizes only two of the leads—the high andground—of a cable, connector and jack
Volume: A general term for a signal's loudness
XLR Connector: A high-quality three-pin audio connector; also called a "cannon connector"; also used for AES-EBU-format digital audio connections
Y cable: An audio cable with one jack on one end, and two on the other

Roland®

Roland Corporation U.S., 5100 S. Eastern Avenue, Los Angeles CA90040-2936

2688US

Mixing tips

Top-flight engineer and mastering king Chris Lewis reveals some tricks of the trade...

The mission is enticing, but just like a busty Bond lovely, it's not all it seems. Go South 007, and meet a man called Chris Lewis who will give you an insight into the 'black art' of mastering.

For not only is this man an acclaimed live and studio engineer, he also masters other people's mixes for a living, and is therefore the ideal candidate to provide The Mix with top tips on how to make tracks more 'radio friendly'. What's more, he does this mastering from his living room, which shows that professional results are possible from home. Hmm... sounds intriguing, M. Accompanied by Q and his array of photographic gadgets, we set off cautiously for our target: MELT 2000 studios.

However, when we arrive at our assigned meeting place, a converted chicken shed that's one of MELT 2000's three studios in their farmhouse complex in deepest rural Sussex, it soon becomes apparent that there's a lot more to this mission than simply a lesson in mastering. Lewis is putting the finishing touches to an MP3 file, which he is 'ISDNing' to Johannesburg for a concert that evening. "It's a female artist called Busi Mhlongo, who I recorded on my last trip to South Africa," he explains. "She is singing at the Kora festival tonight, and she needed a backing track - so I have taken a mix of one of her songs off Pro Tools".

After starting out as a roadie in the early '70s, Lewis graduated to becoming a live sound engineer for bands like Eddie and Sunshine, and Central Line during the punk era. From 1982, a long residency at Ronnie Scott's Jazz Club in London began. In 14 years at the club he engineered and recorded live some of the jazz greats: Art Blakey, Freddie Hubbard, and Clem Curtis, just to name a few, afterwards moving on to projects with Dr. John and Arturo Sandaval, before getting involved in B&W Records (now called Melt 2000) and former owner of B&W Robert Trunz.

"I joined forces with Robert about four years ago and helped him set up this studio. It really was a derelict chicken shed, so we had to start from scratch, installing the equipment and getting Recording Architecture to redesign the mastering room, put in floating walls, soundproofing and so on. The surprising thing is that although you can hear virtually nothing as soon as you leave the studio, we still get complaints from some of the neighbours up the road! I think its just more a case of them not liking muso types in the vicinity, but in terms of noise we have more problem from farm machinery, than they have ever had from us!"

The background to Melt 2000 itself makes interesting reading coming out of Lewis' and Trunz's shared interest in 'world' music, and the fusion of styles from different cultures. "It something that I became interested in at Ronnie Scott's," Lewis explains. "I engineered several Cuban bands during that time and thought they were fantastic". The informative MELT 2000 website (check it out at www.melt2000.com) also documents Trunz's conversion from equipment manufacturing boss to label owner while on a visit to South Africa, with the aim of "reinvigorating modern music by injecting the passion of some of the world's greatest players".

In terms of Lewis' career, this has involved several trips to South Africa, both accompanying MELT 2000 artists on tours and collaborations, and recording new artists for release on the MELT 2000 label. One of the striking things about Lewis' South African projects is the minimal amount of gear he uses, especially when recording in the field. "On my last trip I only took a few flight cases with me," he confesses. "I had a Mac Pro Tools system with a 23 gig hard drive and a Mackie HUI to control it. The only outboard was a DBX 1066 compressor, an ATI mic preamp and a Tascam DA-38, which I barely used. I think some of the bands I recorded were initially really disappointed when they walked in.
They were expecting a huge great mixing desk and reel-to-reel tape machine and there they were confronted with just a computer!"

To many enthusiasts and professionals the likelihood of coming up with successful commercial results from such a minimal set-up would seem to be small. However, Lewis believes that the methods he employs are the ideal solution to the music he is dealing with. "Because all of the people in the bands are such good players the real job is capturing the essence of the performance as accurately as possible. You don't really need loads of effects for

that, perhaps just a little compression. I would say the most important part of my job is miking up the musicians well and coaxing the best performance out of them. Unfortunately, with the advent of electronic music, many of the skills of mic placement are being lost, yet it can make all the difference between a brilliant and terrible recording."

It seems that the man-management element of Lewis' job is as important as any. "Many of these bands have never recorded in a traditional studio before, so anything you can do to put them at ease really helps. For example, when I worked with a band called Amampondo (from the East of the country and supposedly one of Nelson Mandela's favourites) on my last trip we actually recorded the entire album in an old barn, complete with dirt floor. I have also recorded projects outdoors and even in an old British fort from 130 years ago! Recording on location often makes the bands feel at home and therefore their performance sound more natural.

"The other important element in recording in this way is to try not to let the technology get in the way of the performance. Never having been in a studio, some of the 'traditional' bands, especially, find it difficult to record using headphones. In this case I may hook up a pair of out-of-phase speakers behind the musicians instead, to make them feel more comfortable. Another trick is to avoid telling the musicians you are recording whenever possible. For some reason, as soon as they know, the performance immediately takes on a tentative quality. Quite often I will secretly tape the practice runs - which end up being better than the actual take itself."

Listening to some of the South African material, what's striking is how incredibly warm and intimate it is, whilst maintaining the highest standards of recording. It's a quality that's hard to define; this music could have been recorded in a studio yet you simply can't place it there. What's more surprising is the knowledge it was all recorded on Pro Tools. Lewis agrees. "I have become a complete fan because the quality of the Pro Tools DSPs and effects is now so good. Of course I may still record analogue in the studio, but it's no longer a compromise, just different.

The main thing is that you have to watch for digital clipping all the time and run your levels a little shy, whereas you actively want to drive tape hard with the extra warmth you get from a little saturation. However there are so many other things to be gained from using the Pro Tools system. Obviously its mobility, but also the fact that it's so efficient. You can be editing, mixing or striping to silence whilst at the same time recording. It's easy to adjust and very good at drop-ins."

Well aware that we are two hours into the interview and we haven't even touched on mastering yet, we set off hot foot for Lewis' mastering studio: a lounge, complete with three-man sofa and coffee table. Isn't this taking the 'mastering at home' concept to the extreme? "Well it may look like a living room", says Lewis, "but I wouldn't really call it a 'home studio'. There's actually several tens of thousands of pounds of equipment in here, and as you can see, we have also spent money on the room's acoustic design. It's simply a case that Robert and I wanted a comfortable environment in which to master - somewhere that we could be relaxed - and this seemed appropriate."

Again what is striking about Lewis' set-up is the minimal amount of equipment used. Apart from the computer that sits upon the converted coffee table 'rack', there are no more than about 10 pieces of mastering equipment, neatly segmented into analogue devices on the right and digital devices on the left. "I really do believe that a combination of analogue and digital devices is best," explains Lewis. "For example, I often find that analogue EQ is better and more forgiving for the top end of tracks, whilst I will often use digital devices for the bottom end."

In the bottom left of the rack is the studio's latest toy: a Drawmer Masterflow digital mastering processor. "I really like the Masterflow because it's so flexible and easy to use. The display is particularly good because it visualises all the different frequency bands in an intuitive way, allowing you to control crossover points and manipulate bandwidth swiftly. In addition, the 3-band compressor, width control, and valve emulator allow you to process elements of the mix discreetly, say by fattening and narrowing the bass end with the valve emulator and the width control, whilst broadening the top end. I particularly like using these elements to create a tight pumping bass sound on dance tracks.
"By adjusting the crossover points its also possible to start effecting the sub-bass frequencies at the same time, which is a powerful combination. The compression is really powerful too. Even using a ration of 2:1 really blows your socks off! The only drawback with the Masterflow is that it doesn't have a remote control - otherwise it's spot on."

In general, Lewis' mastering methods seem to be very flexible, but he has one golden rule that he is very rarely willing to break: "I tend to avoid mastering projects I have also produced and engineered. It's simply a case of remaining fresh. When you have been that close to a project all the way along, you may miss things that a fresh pair of ears may pick up. So I will tend to take the work to a third party. However, in the case of my most recent South African recordings I'm the mastering engineer as well. I'm aware that my mixes tend to come out a little 'down' - so I like to have someone to bounce off. Luckily having the studio in Robert's house means I always have another set of ears to call upon".

Apart from this one proviso, a mastering project can take any number of forms; "Sometimes the artist or producer will come along to the mastering session, but some people like to remain well clear. I am fairly open to either approach". However it seems that the second method of working can require some additional man-management skills. "Mastering-wise there's definitely a trend for weird effects at the moment, especially with dance music. Some guys have a tendency to want to overdo it and you sometimes have to calm them down a bit."

This draws us onto the subject of mastering at home against going to a professional mastering house. With all the cost-effective equipment that's on the market at the moment surely there is a case for doing the job at home if you are working to a budget? Lewis disagrees. "This studio may be in someone's house but it's definitely not a home studio. In the end you cannot beat the experience of a professional mastering engineer. I have lost count of the number of 'commercial' recordings I have had to remaster because not enough care was taken with them in the first place.

The art of the mastering engineer is to get the recording as loud as possible without killing the dynamics, yet often I receive tracks with levels that are one long red line, so there's absolutely nowhere else left to go. Even worse, some tracks suffer so much from digital clipping that they take on an overly aggressive artificial feel about them".

Before taking our leave we listen to some of Lewis' latest projects including new tracks by Juno Reactor, and Greg Hunter, who mixes traditional Egyptian musical styles with dance music. Both manage to be in-yer-face while retaining a warm subtlety and naturalness that belies the treatment they have received. Both tracks elegantly make the point that the gear may be affordable, but - just like mixing - that's no guarantee that it's going to produce the results you want. Success still remains a matter of skill, subtlety and patience.

Recording Vocals (pt I)

How to get the very best vocals, starting with mic types and other hardware...

Ask people what the most important element in a recording is, and most will agree that it's the vocals. And no matter how good your music is, a sub-standard vocal can kill it quicker than Prince Naseem can duck n' dive.

Whilst they certainly help, top-quality microphones and state-of-the-art recording equipment are not a prerequisite for a great vocal recording. Excellent results can be obtained using more modest equipment. Nor do you need the voice and talent of Frank Sinatra or Aretha Franklin - with a little patience and the use of a few simple techniques, any singer can end up with their best possible recorded vocal.

Over the next few issues of The Mix this series will cover the basics of recording vocals, with techniques, hints and tips to help get the best possible performance recorded with the best possible sound. So let's start from the beginning - what equipment to use, and how to set it up.

Microphones
Both condenser and dynamic microphones can be used to record vocals. Generally in professional studios, large-diaphragm condensers are used, as they have a refined sound with a wide dynamic range and extended frequency response. Many excellent vocals, however, have been recorded on commonly available dynamics like the Shure SM58.

Choice of mic is down to what you have available, but in a situation where you have several different models, make your choice based on which mic suits the singer's voice for a particular song. Many engineers and producers will put up several mics initially to check which one sounds best.

Having chosen a microphone, it's preferable to mount it on a stand. Most mics come supplied with a mount, and the more expensive ones will have a suspended cradle mounting to isolate the microphone from shock and vibration.

It is possible to record vocals using a hand-held mic like an SM58 but, in terms of the sound being recorded, there are several reasons why this is not ideal. Firstly, there is handling noise to consider - the sound of the singer moving their grip on the microphone and moving it around will be picked up. Secondly, unless the singer is very experienced with mic technique, the mic will be held at different distances from the mouth at various times, resulting in small changes in timbre and level. And, thirdly a hand-held mic rules out the possibility of using a pop shield.

Now, having pointed out the drawbacks, it must be said that there will always be some singers who feel most comfortable using a hand-held mic. In this case, recording with a hand-held is the way to go, because a relaxed and confident singer is going to turn in a better performance than one who is uptight about having to stand still and sing into a stand-mounted mic. A slightly less-than-perfect sound is a small price to pay for a great vocal performance.

Arctic rolls
The next thing to consider about a microphone is its polar pattern. Many mics, especially less expensive ones, have a fixed polar pattern, usually cardioid. Others have a switchable polar pattern, but unless you are after a certain effect, switching it to cardioid is preferable.

Cardioid mics are the norm for recording vocals as they accept sound from directly in front and reject much of what comes from the back and sides. This is important in the context of the room where the vocal recording is to take place, as reflections from the walls may be picked up by the microphone, adding some of the ambient sound of that room to the vocal sound. This would obviously be more pronounced if an omni pattern was selected on the mic.

Now, there may be occasions where you will want to record in a particular room to pick up the sound of the room

or the reflections from, say, a window or wall in the room, if that will suit the track you are working on. But in most cases it is probably best to record a vocal in a dead-sounding area and add any ambience at the mixing stage using a reverb unit, because once ambience is recorded with a vocal, you are stuck with it

Some studios have acoustically-treated vocal booths, but if you have to record your vocals in a normal-sized room, a dead area can be created by siting acoustic screens around the microphone. The DIY approach to this for home recording is to hang curtains, duvets, blankets or something similar around the singing area.

A couple of self-assembly bedroom tidy rails from the Argos catalogue make an inexpensive and practical framework to hang material on and construct a functional vocal booth. These can be disassembled and stored away when they're not needed.

6" is ideal
A singer's distance from the mic can make a lot of difference to the sound recorded. A distance of 6" or so is perhaps a good starting point, although experienced singers will work the mic by leaning into it for some passages and moving back for louder sections.

Sing too far away from the mic and more of the room ambience will be picked up; sing closer to the mic and more of the proximity effect comes into play. Proximity effect is a pronounced boost in low frequencies which results in the voice sounding bassier when singing very close to the mic, and it can be successfully exploited by an experienced vocalist.

It is best to try to keep a vocalist at a consistent distance from the mic, particularly when doing multiple takes and where he/she has to leave the booth to listen to playbacks and then go back and sing the odd line. If the same distance from the mic is maintained, variations in volume and timbre between takes is minimised, and dropped-in lines will sound more natural.

Once a singer is at the optimum distance from the microphone, mark the position of their feet on the floor with gaffa tape so that they can go back to the same position each time, and don't forget to mark the position of the mic stand at the same time in case it is accidently moved.

The height of the microphone on its stand in relation to the singer is also a factor to take into consideration. Some like to sing up to a mic suspended a little higher than them, but this can strain the voice if the head, neck and shoulders are stretched up. A mic that is suspended too low is also not ideal if it causes the singer to hunch over, although this at least puts less strain on the neck and shoulders.

An advisable starting position is to have the capsule level with the singer's mouth and then move it if necessary to suit the singer's most comfortable stance. Having the capsule level with the singer's mouth creates its own problems, as it is more susceptible to blasts of air, but there are methods to counter this, the most important of which is the use of a pop shield.

A pop shield is generally put up a couple of inches in front of the mic and its basic function is to stop plosives, which are the popping sounds from blasts of air usually produced by singing the vowels 'B' and 'P'. The pop shield also serves to protect microphones from spit and moisture produced by the singer.

Commercially available pop shields, which usually have a gooseneck and a clamp allowing direct fixing to the mic stand, are fairly expensive. However a home-made substitute can easily be constructed from a pair of tights stretched over a bent wire coathanger - just remember to wash them first if they've been previously worn!

If you cannot attach the pop shield directly to the mic stand, try using a second mic stand purely as support for the pop shield. One useful trick is to fix a pencil vertically down the centre of the pop shield, as this tends to dissipate the energy of blasts of air before they reach the mic. If popping problems still persist, try getting the singer to sing slightly to the side, above or below the mic. If a singer has difficulty doing this and needs to focus directly on the mic, put up another mic that is not plugged in and site it right next to the real vocal mic. Then let the singer sing into this dummy mic.

Whisper to a scream
Microphones have to be connected to a pre-amplifier, and there are two options available. Connection can either be into the mic amp in a mixing desk's input channel, or into a standalone mic pre-amp. These provide a higher quality signal path to the recording medium than that provided by the average mixing desk.

A budget mixing desk will have identical mic amps on all of its input channels and these are built to a price, so a standalone pre-amp, relatively more expensive than one desk input channel, ought to have better quality components and a cleaner signal path. Also, a shorter signal path to the recorder is usually provided by a

standalone pre-amp which can connect directly to it, whereas the signal through a mixing desk may have to pass through input channel, group busses and patchbay.

Compression is near-essential to even out the performance when recording vocals. The human voice has a huge dynamic range (from a whisper to a scream, to use the old cliché), and a compressor will 'squash' that range a little. Don't go over the top, though; reducing the peaks by a few dB ought to be sufficient. Once compression is recorded you can't take it off, so it's best to err on the side of caution. More compression can, of course, be added as needed at the mixing stage.

Several of the standalone pre-amps on the market have their own compressor built in. When recording through a desk's input channel, a compressor should be connected via the channel's insert points.

EQ can also be applied when recording vocals, perhaps to remove a bit of nasal honk from a voice, to brighten up the sound a little, or, most usefully, to filter out some of the bottom end of the spectrum. Real low frequency sounds, such as outside traffic rumble or the sound of the singer's feet moving on the floor can be transmitted up the stand to the microphone, and an increase in bass due to the previously mentioned proximity effect can also be a problem.

To get around this, switch in a high-pass or bass roll-off filter. Most mic pre-amps and desk channels, and some mics, will have a switchable filter operating at somewhere between 75Hz and 100Hz, cutting out most of the low end below that figure.

EQ should, however be applied with caution. Adding too much top-end boost, for example, can often exaggerate the sibilance of the voice. It's best to record a vocal flat, but if you feel the need for EQ, use it sparingly. And while you may be tempted to use a noise gate or downward expander to cut out noise between phrases, our advice is not to. It's too easy to chop the end off notes and make the vocal sound unnatural.

Processing of this sort should be left to the mix stage, when time can be taken to set it up accurately.

Condenser or dynamic?
Although there are other designs, the microphones most commonly used in studios today fall into one of two categories - condenser or dynamic. A microphone is simply a device which converts acoustic energy (sound waves) into electrical energy, and the dynamic and the condenser each do that in their own way. This has consequences for the sound produced, and hence the use to which each is put.

A condenser mic, also known as a capacitor mic, has a thin diaphragm that is supported around its rim at a small distance from a thicker backplate. The theory is that the two form the two electrodes of a simple capacitor, and are oppositely charged by the application of a polarising voltage. When the diaphragm moves in response to sound waves, the spacing of the diaphragm and backplate (and hence the capacitance) will vary, and this is used to generate the output voltage.

Because a voltage has to be supplied to the backplate and diaphragm, a mic of this nature needs a power supply. This usually comes in the form of 48V phantom power supplied from the mixing desk or mic pre-amp. Condenser mics are more difficult to manufacture than dynamics and are therefore more expensive; they are also not as rugged and are more susceptible to changes in atmospheric conditions, so should be stored, and handled, with care.

In use, a condenser is generally more sensitive than a dynamic and has a better transient response. It also has a wider frequency response, so can pick up more top end than a dynamic, making it very useful for instruments like cymbals, acoustic guitars, and vocals.

Condensers can be built with two diaphragms, and by changing the voltage of the second diaphragm in relation to the first, the mic is capable of several different polar patterns - from omni-directional through cardioid to figure-of-eight. Some condensers are designed with a valve in the circuitry; these do not need phantom power as they usually come with their own power supply. Valve mics provide a different tonality than the standard condenser, with an added warmth in the sound.

Another variation from the standard condenser design is the electret mic, which uses a permanently charged electret material to charge the capsule. These mics are usually cheaper than condensers and can often be run from a battery if you do not have a phantom power source.

Dynamic mics work because of the electromagnetic interaction between the field of a magnet and a moving coil conductor. A coil of wire, surrounded by magnets, is fixed to the back of the diaphragm, the motion of which results in the coil cutting through the magnetic field, inducing an electric current in the coil.

Unlike condensers, dynamic mics do not require any power supply. They are more robust, and can cope with high sound pressure levels.

Because dynamics are pressure- operated, their polar response can only be either omnidirectional or cardioid, and most handheld dynamic vocal mics are cardioids. Dynamics are also limited in their high frequency response, some having an upper limit of 16k (a good capacitor will go up to 20k).

Mics designed for stage use will often have a bass end roll-off built in to counteract the proximity effect, and many have a presence peak built into their frequency response somewhere up around 5k. This is designed to help vocals cut through a mix. Some very well-known rock singers rcord their vocals with dynamic mics for that particular punchy sound.

Microphone Polar Patterns

There are four basic options when considering a mic's polar pattern:

A **cardioid** (or unidirectional) mic is so named because of its heart-shaped response. It will pick up sound mostly from the front. Dynamic cardioid microphones are popular for vocals because of their off-axis exclusion, and robustness, but condenser cardioids are much better for the studio vocalist.

A **figure-of-eight** microphone picks up sound from both front and rear of the diaphragm, but because the opposite sides are out of phase, side-on sources get cancelled out. Figure-of-eight microphones have the potential for very accurate recordings.

A circle is the polar response of an 'ideal' **omni-directional** microphone. In practice, the response favours the 'open' side of the capsule at higher frequencies, so off-axis sources can be dull. Omni mics are particularly resistant to wind and handling noise.

A **hypercardioid** microphone mixes the cardioid and figure-of-eight patterns to produce a 'thin' cardioid with an out-of-phase area at the rear. Because of this, the hypercardioid is good for reducing the effect of reflected or off-axis sounds, such as room reflections.

Vocal Mics

£2,000 +
Some 1950's valve vocal microphones are still in use today. The Neumann U47 and the AKG C12 are universally regarded as classics and sell for well in excess of £3,000 on the used market. AKG now produce a reissue of the C12, in the form of the C12 VR, M149, and the soon-to-be-released M147.

Below £2,000
The Neumann U87 and AKG C414 are the two most common vocal condensers used in studios today.

Below £1,000
Anyone wanting to buy a decent condenser under £1,000 for vocal use is spoilt for choice. AKG's SolidTube incorporates a valve in the design. Beyer's 834 and Audio Technica's 4033 and 4050 are all respected, and many of the eastern European imports give great results for a reasonable price. Australian-made Rode Microphones represent excellent value for money, and AKG weigh in with several of inexpensive contenders: the C4000B, C3000, C2000 and C1000S.

Below £100
There are loads of inexpensive dynamic mics available, but think of a dynamic vocal mic and you invariably come up with the Shure SM58. This rugged workhorse is the industry standard hand-held stage mic, but its partner, the SM57, will also give good recorded results.

(In part II, we look at setting up a headphone mix, and getting the best out of a singer.)

Trevor Curwen The Mix

Recording Vocals (pt II)

Tips on creating an environment that will make a singer shine...

Last month we looked at the mics and other front-end gear that you need to have in place before you can even start recording the ultimate vocal take. Now it's time to bring the singer into the studio, and as an engineer/producer, there's much you can do to get them to give the best possible performance.

Here's a question: at what point in creating a finished recording should the vocals be recorded? Now if you're dealing with the sort of band that likes to record everything live in one take for that warts-and-all sound, you don't have to decide. Obviously the vocal would be recorded at the same time as the rest of the band, either in the same room (separated perhaps by screens), or in a separate booth.

More commonly, though, a band will record a backing track with the singer putting down a guide vocal to be replaced later by an overdubbed vocal. Or a song will be built up track by track, with one overdubbed or sequenced part after another, and the vocals will be recorded at some point in the process.

There are a couple of different approaches to this. Some singers will prefer to do their vocals after all the instrumental parts have been recorded. In one way this is illogical, as it may put undue strain on the singer - he/she may only have one free track to record on, and will and have to wait until the end of the session to sing.

The other approach, which is the one I favour, is to record the definitive vocal as early as possible in a song's recording. This approach allows vocals to be recorded when there are plenty of tracks left free for alternate takes, and allows the singer to record vocals as and when the mood takes them, putting them under less stress.

The resultant vocal performance should also inspire a more sympathetic performance from other musicians in putting the finishing touches to a recording, and will give a better sense of what other parts need to be added for a finished production.

Certain things have to be in place, of course, before a vocal can be recorded. The backing track should be together enough for the singer to be able to relate to it and get the vibe of a performance, and there should be enough melodic instruments on it for the singer to pitch his or her voice against. It's not impossible, but it really is quite difficult, to pitch a vocal and keep it in tune against just bass and drums in the headphones, which brings us to the subject of...

Headphone mixes
Headphones can be categorised as either open or enclosed and, while it is possible to use either design when recording vocals, the closed design is generally the preferred option as less sound escapes into the microphone. Any bleed from headphones will be picked up by a sensitive vocal mic, and in the worst cases can cause howl-round type feedback (watch out for this when a vocalist with a loud headphone mix takes their phones off near the microphone).

A little bit of backing track bleed is no great problem, as it is usually only noticeable when the vocalist is not actually singing and can be gated out of those quiet sections at mixdown if necessary.

The overall level of the headphone mix will vary from singer to singer and depends to an extent on what elements are in it, but a good rule is to start out with the level fairly low and bring up the volume until the singer is comfortable with it. Then do the same with the talkback level. An identical set of headphones on the same feed as the vocalist's is useful for the engineer in the control room to check the general mix and levels.

The headphone mix should be sent out from the desk on an auxiliary send that is set to pre-fade, so that it will remain constant regardless of any monitor fader adjustments made in the control room. Each instrument should be fed into the mix from its monitor channel.

The singer will need to hear some of the elements of the song that have been already recorded, and these should be put into the headphone mix. It's not necessary to put absolutely everything in the headphones, just select whatever the singer needs to turn in the best possible performance.

Sometimes it may be necessary to add a few things that are not going to be used in the finished song, perhaps a series of clicks as cues for when various sections start (Q. How do you know when a singer's at your door? A. He never knows when to come in) or a sustained keyboard note to help pitching at the start of a phrase. If you're running a sequencer alongside your recording medium, these things can be quickly input and muted if not needed.

It might be useful to add a little reverb to some of the instruments make the sound more polished, and to add a little reverb or delay to the singer's voice to help them out. Through a set of headphones, a singer isn't hearing their voice as they hear it in the real environment - when we sing or speak, we hear our voice as a combination of the external sound in the air and internal bone convection, which is why hearing your own voice back from tape is very different from how you perceive it normally. Wearing headphones obviously changes the relationships in the way we hear things, so reverb may help put the 'air' back around the voice.

Because of this, some singers like to tilt their headphones, using only one earpiece while leaving the other ear free to hear the natural sound. If this is the case, be careful of bleed from the unused earpiece - you can eliminate sound from it altogether by panning the sound or turning off the feed to that side.

The actual level of the voice in the headphones in relation to other things can be crucial to a performance, especially when it comes to pitching. A degree of experimentation with the level will probably be necessary before you arrive at the optimum mix for the vocalist. It's not the case for everybody but too much vocal in the headphones can tend to make the singer go flat, while too little might cause them to push things a bit and go sharp.

Some vocalists may not like using headphones at all, and might just want to sing while listening to monitor speakers. There is a way to do this that minimises the amount of spill picked up by the microphone: let the singer face the monitor speakers singing into a microphone that is equidistant between the speakers.

Reverse the wires at the back of one of the speakers, putting them out of phase, and switch the backing track to mono. With this method, the backing track tends to cancel itself out and a minimal amount of spill is picked up.

In the mood

A great vocal performance is not just one where the technical aspects of timbre, tuning and timing are perfect; it goes beyond that to create something powerful, skilful and moving that really connects with the listener. As a recordist you can't provide a vocalist with talent, but you can help create a mood, atmosphere, and environment that will let any singer concentrate on giving their best possible performance.

Recording vocals can be a traumatic thing for some singers, particularly ones who have never recorded before and are suddenly confronted with the strange, naked, sound of their own voice coming back at them through the speakers, so you need to make things as stress-free as possible for them.

Even professional singers will often want to convert a vocal booth into their own private sanctuary with various combinations of pictures on the wall, pot plants, flowers, Persian rugs, coloured lighting and joss sticks. Whatever makes them feel relaxed and able to perform is fine.

Different singers will have their own methods of getting in the mood for singing. Some may like a drink or a smoke, and there are tales of some who like to strip off and sing, and, of course, the infamous story of The Doors' Jim Morrison recording a vocal while allegedly having his todger serviced by a young lady on her knees in front of him.

Of course there's no need to provide that sort of service in your own studio; a glass of water usually suffices for most singers.

Where's the sauna?

The first basic priorities for creating the right atmosphere are heating and lighting. A neutral temperature is preferable, and subdued lighting has to be a much better option than the glare of a fluorescent striplight. Some singers even prefer to sing in the dark.

Different singers will have different reactions to people watching them record. This can range from wanting to have loads of people around urging them on to a great performance, almost like a gig, to not having anyone

around at all. On one occasion, with a particularly shy singer, I actually set up a microphone in the control room, showed the singer how to drop in and out of record, and went shopping for a couple of hours while he got on with it. Most singers fit in somewhere between these two extremes.

Consider the situation where a singer is recording in a vocal booth with direct visual contact with the control room, either through a window or via a video link. It can be very disconcerting to them to see people laughing - remember that they can't actually hear the control room banter unless the talkback is on, and they might think that everyone is laughing at their out-of-tuneness or that naff line in the lyrics that nobody has heard clearly before, when in fact all that has really happened is that the keyboard player has dropped a particularly choice fart.

So, if the situation calls for it, clear the area of all but essential personnel. A curtain in front of the vocal booth or area that will cut visual contact in both directions, which the singer has the option of pulling across, is also a good idea.

A singer who plays guitar on stage but is in a booth simply to do vocals may feel a bit strange without their instrument. The solution to this is simply to let them hold the instrument while singing - this might seem obvious, but is something that is easily overlooked.

The human voice can get worn out by a long singing session, so it's important not to overdo things. Fortunately the voice is very resilient, so a rest between sessions will restore it. Warming the voice up before a session is a useful exercise, and a singer should be given the opportunity just to go somewhere and run through a few things so that they are not going straight into the recording cold.

Mornings don't seem to suit vocalists, the reason may be physiological - something to do with the larynx opening up as the day goes on - but the important thing about the timing of recording vocals is to catch the mood and do them whenever the singer feels he/she is ready.

On a long, ongoing session it makes sense to have a microphone set up so that you are able to do vocals whenever the singer feels like it. If you have to fit everything into a one-day studio session there isn't a lot of scope available, but on an album session over the space of a month there will be days when the singer doesn't feel like singing or their voice is shot, and there will be days when they will be really up for it. Flexibility is the key.

Scheduling the vocal for the end of a session with a specific amount of time set aside puts pressure on the singer. They'll have to wait around through the whole of the preceding session, and there's always the chance that they might have a sore throat or some other ailment when the time actually comes round.

Communication between the recordist and the singer is usually carried out via the talkback button, and it is important to maintain that communication. Getting on the talkback straight after a take lets the singer know you have been listening and doesn't leave them standing in silence wondering what to do next.

It might be the case that the singer wants to make several passes at the song, one after the other, on different tracks, so it is important to work quickly in getting another track ready and not leave them waiting while you fiddle with adjustments, as this may break their flow and concentration.

That was shite!

Advice and criticism about the vocal depends on the recordist/engineer/ producer's relationship with the singer. Experienced producers have their own particular ways of helping improve a singer's performance, whether by encouragement, coercion, or even abuse, and will be able to read the particular situation and react accordingly. One helpful tip is to make criticism, if solicited, fairly specific. Few singers will want to hear that they can do it better but they might want to hear about a particular word in the second verse that they need to pitch slightly differently.

Some singers will know exactly how good they are and when they've got a good take. Others will not be so sure, and will need a little positive feedback. It's good form for the recordist to have a copy of the lyrics so they know exactly where they are in a song, and to keep a notepad handy to jot down comments on each take.

And if this article makes it sound like all singers are a bunch of prima donnas, that's not the case at all. Many singers will just get in the booth, regardless of anything else, and sing their hearts out.

Nevertheless, a few of the tips here will go a little way to helping any singer turn in a great performance.

White Lies: Your guide to producer speak...

They say: "I'll run the tape and get a level"
They mean: "I'd better record this as you're obviously only good for one take."

They say: "That was brilliant but let's do another track anyway."
They mean: "Bloody hell, that was crap. Let's hope you do better next time."

They say: "I'll turn up the piano a bit more to give you a better feel of the track."
They mean: "Have you ever tried singing in tune?"

They say: "Would a click track be useful?"
They mean: "Can't you fucking sing in time?"

They say: "Just give it a bit more projection."
They mean: "This sounds boring and lifeless."

They say: "Your intonation was a bit under."
They mean: "You were flat."

They say: "Sing more from your chest."
They mean: "You're singing like a strangled cat."

They say: "Shall we go for a quick pint?"
They mean: "Will you loosen up, please?"

They say: "The tuning and timing were a bit iffy."
They mean: "That take had no redeeming features whatsoever."

They say: "One more for me."
They mean: "I've run out of things to say without repeating myself."

Trevor Curwen The Mix 02/99

Recording Vocals (pt III)

In part III we share some secrets on the art of comping

If you've read the previous two parts to this feature you should have all your hardware in place, and your vocalist will be suitably chilled-out and raring to give it some into the microphone.

This time we'll look at how to construct a definitive lead vocal from several takes, as well as double tracking and backing vocals. So let's hit 'record' and roll through the song.

Different singers and producers will approach recording a vocal in different ways. Vocals can be sung a whole song at a time, a section (verse or chorus) at a time, a line at a time, a word at a time or even a syllable at a time a method that allegedly came as something of a surprise to the Pet Shop Boys when working with Dusty Springfield.

Ultimately the method used has to be what the singer is most comfortable with or is prepared to do, whether that means concentrating on one line at a time, singing it over till it's exactly right and only then moving onto the next one, or perhaps singing the whole song in one take and going back to repair any duff phrases or words by dropping in.

Any method or combination of methods is valid if it gets the job done. A situation where there is only one track left to record the vocal on is always going to be more stressful than having several tracks available.

If you only have one track to work with, getting drop-ins accurate is essential and you have to hope that any dropped-in vocals are going to be an improvement on the ones you are erasing, because you can't go back to it if they aren't.

With more than one track available more options open up; notably the fact that you can keep a vocal on one track and record another one on a different track.

My own preference is, if possible, to work across three separate tape tracks while keeping another free to bounce the best bits of each across, to create a vocal that is hopefully greater than the sum of its parts.

Another personal preference, if the singer is willing, is to record a full pass of the song each time. This gives the singer a chance to warm up and get into the feel, momentum and dynamics of a song, which ought to produce more of a performance than would be achieved by constantly stopping and starting the tape.

However, this isn't an approach that will work every time. Some singers will prefer other methods and their needs are paramount there is nothing to be gained by coercing people into doing things that don't feel right for them.

In a situation where you are recording a vocal across three tracks, it's possible that a definitive vocal might be captured solely on one of them, negating the need to make a 'comped' vocal.

However, if bits of several of the tracks are to be used, a consistency in sound from track to track is pretty essential as level and timbral changes will make it hard to match the sections up. To achieve this, keep the vocalist the same distance from the mic for all the takes, (gaffa tape on the floor to mark the position will do the trick), and once you have settled on a headphone mix, amount of compression and EQ, stick to it.

So let's go through the method of working with three tracks and constructing a composite vocal. We'll be refering mainly to working with a conventional multitrack recorder and mixing desk here, but anyone using a computer-based system will still find the following procedure of use.

Recording it
If you're lucky, the first pass a singer takes at a vocal might just be the perfect performance. It does happen the

adrenaline kicks in, the singer puts in the performance of their life and couldn't possibly do it better. It's usually fairly obvious when this happens, and a decision has to be made whether to go for a second take on a different track anyway.

If you do go for a second take, it will soon become apparent whether or not it's going to top the first. But the best performance doesn't often happen on the first take. It's more common for a singer to have several attempts at a song, improving with each take. In this case, take track 1 out of record-ready and record another performance on track 2.

With some singers there is a palpable sense of relief at having got that first vocal in the can. It allows them to relax a bit and the second performance benefits from this. Another thing to note is that singers do need to warm up, both in terms of of the voice getting physically stronger, and of the performance becoming stronger through repetition. So it is likely that track 2 will be an improvement on track 1.

To record a third performance, take track 2 out of record-ready, mute its playback channel, rewind and go for another take, this time on track 3.

At some point in this process the singer may want to hear some or all of the performances back. That's fine it will give them a chance to hear what they have been doing and assess how they can improve on it, as well as giving them a break, although this has to be balanced against losing the natural flow and momentum of the session.

Personally, as engineer or producer, I'd rather have a singer build up a head of steam and do a few takes in quick succession before coming in to listen, but each session and singer is different and working methods must be adapted to the prevailing circumstances.

If the performances are getting better with each take and the singer is still game for it, then the next step is to record a fourth performance or take. Note the use of the word 'take' and not 'track' at this stage it is possible to move onto a fresh track and record another vocal, but having different vocal performances spread over, say, four, five or six tracks makes things a bit unwieldy when it comes to listening back and choosing the best bits.

If occasion or the singer demands it then more tracks can be used, but working on three tape tracks only is a much simpler and much more workable proposition.

So, sticking to working with three tracks only, take number 4 will usually be recorded over take 1 on track 1, assuming that the singer agrees that the first take was poor compared with the last one and that the performances are generally improving. So here we go, with playback from tracks 2 and 3 muted and recording on track 1 again.

At this point it's worth noting that a tracksheet, pencil, rubber and notepad are essential for keeping track of everything, especially a note of which take is the best of the three at any point.

From this point on, as long as the singer's performance continues to improve, the process can continue with a fresh vocal being recorded over the worst of the three already on tape until the performances reach a peak, or the singer decides they've done enough. Use your judgement to tell you when you've got enough good vocals.

At some point it is important to draw a line under the proceedings, as vocals will not keep improving indefinitely and voices get tired.

When there are three good performances recorded, it's time to assess what you've got. It's just possible that one of them might be the definitive performance in itself, but if that's not the case, bits from all three performances can be combined to create the best possible comped vocal.

Appraisal

As an aid to appraising the recorded vocal performances, take a large sheet of paper and make up a chart with the lyrics line-by- line down the left hand side of the page, and three vertical columns on the right.

Next, sit down with the singer, listen back to each performance, give each vocal line a score out of three and mark it in the vertical column for that track ticks or crosses are a good visual indicator. It's also a good idea to jot down quick notes of odd words that may spoil an otherwise good line, or that stand out in an otherwise crap line.

After playing back all three tracks of vocals you will have a fairly good idea of which lines should make up the final edited version of the lead vocal. It may be the case that one track stands out from the others, with most lines scoring threes, and that line should form the basis of the composite track, with any imperfect lines or words replaced by good ones from the other two tracks.

There are obviously several routes through the song and the important thing is to run through the song a few times and, by switching between tracks with the channel mute buttons, try out the various permutations to see which sounds best.

You may find that you have to switch tracks in the middle of a line, but eventually the best-sounding route through the song will be found. Mark it on the chart with a highlighter pen, taking care to note the exact position in each line where the tracks should be switched.

But what if you find yourself with three good vocal takes, all with a real clunker of a word, or a line that is equally bad on all three tracks? The obvious solution is to get the singer back in the vocal booth and attempt to drop in a good version of the offending word or line.

The same method as before can employed, utilising all three tracks if required. Alternatively and this only possible where the same word or line appears twice in the same song the word or line can be sampled and dropped in where needed.

Anyone working on a digital system with good editing facilities will have no trouble moving the odd word or line around, but for anyone using tape sync'd to a sequencer, the easiest way to sample a vocal line and move it to another part of the song (without lots of editing of start times and so on) is to initiate sampling using a MIDI note at the start of the bar, and sample a whole block of vocals.

Moving the line to another position in the song is simply a matter of moving the same MIDI note to the bar where the new line should start, and using it to trigger the sample.

Bouncing

Having decided which lines will make up the best possible comped vocal, it can now be put together, and depending on the medium you are working with, the deed can be done in different ways. Anyone with a computer-based digital system will have their own method of editing a vocal together; anyone using a digital or analogue tape-based system can bounce bits from the three tracks across to another track.

With analogue tape, any track bouncing is going to slightly degrade the sound quality. This may be unacceptable to some, but the advantage is that once the composite track is recorded the original vocal tracks can be erased, effectively freeing up two tracks. If there is no shortage of tracks, then the composite vocal can be constructed at mixdown by switching between the three no bouncing required.

This is straightforward if your console offers fader and mute automation. Otherwise it's a lot of work, but if that's the preferred method it is still useful to make up a composite vocal on one track to be used as the reference vocal for additional overdubs.

To make the comp when working with a multitrack tape machine, the three recorded vocal tracks (source tracks) should be bought back through three adjacent desk channels (adjacent for ease of switching) and routed to a clean track (destination track).

Let's call the destination track '6' for sake of argument (remember never to bounce to an adjacent track on an analogue machine). The three source tracks should not be routed to the main L/R mix, but the destination track should be, as this is the one we need to monitor.

With the destination track in record-ready, set the tape running and switch between the three source tracks by using the desk's channel mute buttons and following the previously drawn-up chart. If the three vocal tracks have been recorded consistently they should all be set at the same level, but it may be that some minor level adjustments have to be made for the odd word or phrase, and a piece of masking tape fixed alongside the channel fader is useful for making the various level changes.

Sometimes a little EQ may be needed to even things up on one of the tracks where the singer's timbre has changed, and if you decide that a little overall EQ or compression should be added to the composite track while it is being committed to tape, then outboard processors can be patched in on the group insert.

Once you're well-versed with your fader moves and channel switchings, roll the tape and record onto your destination track. If you make a mistake, just roll back and drop in. When you have finished, check the composite track for glitches, level changes or anything that doesn't fit, and if a section is not good, simply roll back and do that section again.

Only when you are completely satisfied with the finished vocal is it time to erase the original vocal tracks 1, 2 and 3, and if you are really paranoid about losing anything put them on a safety DAT first for sampling later vocal on one side of the stereo, timecode on the other.

Double Tracking
Once a definitive lead vocal is in the can, it's time to look at recording additional vocal parts. A lead vocal can often be enhanced by double tracking, which is basically doubling the vocal part and then using the two vocal parts mixed together as the lead vocal.

Double tracking can add a certain depth and thickness to the sound of the voice, and is a technique often embraced by those singers who don't particularly like the sound of their own voice or consider it to sound weak on its own.

Automatic double tracking or ADT is an electronic method of achieving a double tracking effect using delay, and although this is a very valid and effective sound in its own right, it does sound different from real double tracking. With ADT the doubled vocal is a clone of the original vocal, but with real double tracking the slight differences between the newly recorded vocal and the original can give interesting results as the two voices blend together.

For good results when double tracking, the doubled vocal must closely follow the original, so it's important to hear the original at a reasonable level in the headphone mix. Any problems mainly arise from timing namely coming in early or late on a word and not matching the tails of words precisely.

This is most apparent on hard consonants like 'T's and on sibilant sounds. Slight timing differences on these can cause a smearing of the sound, but even if the timing is correct, problems can occur with the stacking up of hard consonants or sibilants producing a peak of energy in the vocal. This is particularly noticeable when reverb and delay are being used on the vocal.

One way to counteract these types of effects is by softening some of the words when singing the doubled vocal part. An example might be that if a word has a 'T' at the end of it, it may be a good idea to ease off on that 'T' or not to pronounce it at all when recording the second vocal.

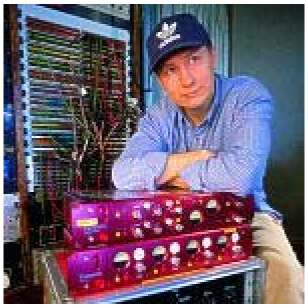
The two vocals might blend together more easily with only the 'T' from the original being sufficient to carry the dictation of the word. This is a technique that takes a little practice, but by running through a song a few times it will soon become apparent where to soften the second vocal and perhaps shorten the tails on some words.

This softening or blurring of the words is not just something that can be done on double tracked vocals. Backing vocals and stacked harmonies can often benefit from this, and in some cases it can help backing vocals blend more easily into the vocal track and not clash with the lead vocal.

Another thing often worth a try when doubling vocals and tracking up harmonies, especially if you are doing all the harmonies yourself, is to varispeed the tape slightly when recording. This obviously means you will have to sing slightly sharper or flatter (depending whether you take the speed up or down) to match the already recorded sound, but when the speed is returned to normal, the newly recorded voice will have a slightly different timbre to the original, which will add some variety to the sound.

Trevor Curwen The Mix 03/99

Recording Vocals (part IV)



We finish off our series with a look at mixing and processing...

The previous three articles in this series have all been concerned with getting the best possible vocal performance recorded with the best possible sound. In this final installment, the focus is on techniques that will make that vocal sound as good as possible blended in with the other instruments in the mix.

The lead vocal is usually the focal point of a song and, as such, will be a fairly prominent feature in the overall mix. However, it should still sound like a coherent part of the track, not like it has just been stuck on top of the other instruments.

There are ways and means of achieving this. When constructing a mix there are four basic elements that can be adjusted for any particular sound. These are Volume, EQ, Panning, and Effects, and all are important in setting a vocal in its appropriate context.

The overriding element in the whole thing might just be the overall volume of the vocal, and although rough volume levels will be set as the mix progresses, final level adjustments might not be made until the end of the mixing process. In fact, some producers will do separate 'vocal up' and 'vocal down' mixes (with the vocal at different levels), which gives the record company different options and may save having to remix the whole track. Before the final settings of the fader on the vocal channel are decided, however, there are other elements to be addressed...

Compressors and gates

Vocals are usually recorded with compression, but not too much, as once recorded it can't be undone. The mix stage is the place to add more of this effect, and there is plenty of scope to pile it on and see what works best. Compression will make the overall level of the vocal more consistent and allow it to be heard clearly in the mix.

It may be that there are various unwanted breathing noises, grunts or headphone bleed in the gaps between the vocal phrases, and any compression applied is likely to bring the level of these up. If any such noises are causing a problem, they can be removed by muting the desk channel between phrases, or by gating.

Some compressors, and many of the new voice channel-style processors, will have a built-in expander/ gate. This often takes the form of a downward expander, which lowers the level of any signals falling below a user-defined threshold. These can be very useful if set up to work unobtrusively. Care should be taken to set them up so that the starts and ends of words are not chopped and made to sound unnatural.

The same caution should be applied when using a conventional noise gate to clean up a vocal track. A few run-throughs of the song should allow you to set the optimum settings for the threshold, attack and release controls to be achieved.

Noise gates can also be used to keep backing vocals tight. You may have several tracks of backing vocals that are all meant to start and finish at the same time, but which are actually a bit raggedy. These can be fed through a gate, and a split from the tightest-sounding one can be used to key the gate so that all the starts and stops are closely synchronised.

EQ

The human voice is a natural sound that everybody is completely familiar with, so any over-use of EQ will be easily apparent. Now, if a special effect (like the old trick of making the voice sound like it is coming down a telephone line) is required, a good old fiddle with the EQ knobs might be just the thing. But under any other circumstances, sparing use of EQ (or even none at all) is the best option.

In situations where the sound of the vocal has not been recorded well in the first place, or where the vocal needs a little tonal separation from the other instruments in the mix, there are several areas of the frequency spectrum

which can be addressed.

The vocal may not have been recorded with a high-pass filter in place to remove unwanted low frequencies. In this case, these can be removed at the mix stage with the filter on the desk channel. Anything below 60Hz isn't really going to be useful information in a voice, and removing some of the frequencies just above this (up to about 100Hz), can help to increase clarity without losing the body of the voice.

Cutting out some of the lower midrange can lose some of the body of a voice, but this is sometimes useful for taking out a bit of muddiness. If the voice does have a little too much 'mud' in it try cutting at around 300Hz.

Some voices may have an irritating harshness, either from the voice itself, or from the recording having been made with a cheap dynamic mic. This harshness can usually be located in the 2K to 4K frequency range and can be cut accordingly.

Boosting the top end of the voice a little can open up the sound and let the vocal through. Anything over 5K counts as top end, but make small amounts of boost the order of the day, and be aware that boosting the top end can increase the sibilance in the voice. Sibilance usually occurs somewhere in the range between 5K and 9K but this depends on the particular voice. As a useful rule of thumb, if your EQ has a bandwidth control, cut over a narrow range and boost over a broader range.

Panning
The obvious and traditional place for a lead vocal in the stereo spectrum is dead centre, and this is where you will find it on the majority of commercial releases. However, the taste police won't come round your house to arrest you if you place it somewhere else.

Having the lead vocal dead centre does give it a certain focus, though, and leaves the door open for placing backing vocals in stereo each side of it. In the case of a double-tracked vocal, the two can be panned slightly left and right of centre if desired.

In earlier parts of the series we recommended that vocals be recorded dry (that is, with none or very little of the sound of the room picked up with the voice). Now, at the mix stage, some ambience in the form of reverb or delay can be added to the voice to put a sense of space around it, and help it to sit in with the rest of the instruments.

The amount of reverb and/or delay added to the voice will determine its perspective or depth in the stereo picture. A completely dry voice can sound unnatural but will also sound very forward and intimate, while adding larger amounts of reverb and delay will have the effect of pushing the voice further back into the picture. Aim to get the voice sounding in the same space or just a little in front of the instruments. This will give a certain coherence to the track.

Reverb
Traditionally, reverb was produced by an echo plate: an electro-mechanical device consisting of a large metal plate driven into vibration by a speaker coil. Transducers picked up the vibrations, which were then amplified to produce the reverb sound.

Plates are still in use in many studios and produce a familiar sound that is good on vocals, and which is simulated in the 'plate' settings of digital reverb units. Some reverb units will have specific 'vocal plate' programs, and these can be a good first choice when choosing a vocal reverb, although room and other program types work equally as well.

The amount of reverb used is probably a more critical factor than the type of reverb program. Be aware that bright-sounding reverbs can emphasise sibilance in a voice, and longer reverbs can reduce intelligibility by swamping the sound. Most reverb and multi-FX units will allow editing of the most useful parameters, such as the length of the reverb decay, and the pre-delay which provides a gap between the dry sound and the onset of the main body of the reverberation, so that the dry signal can stand out more from the reverb.

Backing vocals can usually be given a little bit more reverb than the lead vocal to sit them further back in the mix and, if available, a less bright reverb emphasises this effect. Some multi-FX processors will allow a combination of effects to be set up - placing a chorus effect or a bit of mild detuning from a pitch-shifter before the reverb can thicken it somewhat, which can be useful on massed backing vocals.

Delay
Delay can be used on vocals both as a definitive, audible effect, and as a more subliminal effect that helps bed the vocal nicely into the backing track. The type of effect obtained depends mainly on the length of delay used.

A very short delay (below 30ms) can be added to the vocal to slightly thicken it. When this is used, the human ear can't really identify the two discrete sounds (dry vocal and delay) and will perceive the whole vocal sound as being thicker. Slightly longer delays (in the range of 30-60 ms) are used to produce the ADT (automatic double tracking) effect we mentioned last month. In this range, the human ear will distinguish two separate voices and it will seem like the voice has been doubled. Introducing a little modulation into the delay adds variation to the sound.

Another effect achieved by using a relatively short delay time is slapback. This is one repeat, which sounds like a definitive echo, and can be heard on some of Elvis Presley's recordings and on old rockabilly records. Choosing the length of the slap delay is a matter of taste, but try something around 90 ms as a starting point, and as with all delay sounds, experiment with the relative volume of the delay against the dry sound to see what sounds best.

Using longer delays than those in the three types of effect already mentioned gets us into the realm where delays can be set up in time with the track, and these can be extremely useful in bedding the vocal in with the rest of the backing track.

Tempo-related delay
If the tempo of the song in BPM is known, then it is easy to set up a tempo-related delay using a digital delay line (DDL). Delay charts are available that give the correct delay times to use for any given BPM but the times can be worked out anyway by using a simple formula. Dividing 60,000 (the number of milliseconds in a minute) by the BPM gives the delay time in milliseconds for 1 beat or 1/4 note (60,000/BPM = delay time in milliseconds per 1/4 note).

So for a BPM of 120, the formula gives us a 1/4 note delay of 500 ms. To get an 1/8th note delay this figure can be halved, giving a time of 250ms, or it can be multiplied by 3/4 to give a 3/16 delay of 375ms.

If the song's BPM is not known, a rhythmic element in the backing track (like a snare drum) can be fed into a DDL and the delay time adjusted until the delays are in time. Any variations on the displayed delay time can then be worked out mathematically as before. Many of the newer DDL's and multi-FX units now, of course, have a tap tempo button which can certainly speed up the editing process.

Once the tempo-related delays have been worked out, they can be tried out on the vocal to see which of the delay lengths is most effective. One useful combination using a stereo delay is to have a 3/16 note delay on one side of the stereo and either an 1/8th or 1/4 note on the other. Add these to the mix and they will help the vocal sit nicely in the track.

Now, these delays do not necessarily have to be heard above the rest of the track - they work well as a subliminal effect and even if you can't hear them, they do help to keep a vocal sat in the track.

Some tempo-related delays that you really do want to hear can also be applied to a vocal. One popular delay effect is where the occasional word, or maybe the last word in a line, is repeated. There are a couple of ways to achieve this: the first is simply to momentarily turn up the aux. send to the delay for the particular word that needs repeating.

This can easily be achieved with an automation system in place, but it can be a tricky thing to get right each time if done manually. A favoured method is to assign or split the vocal to two desk channels: one for the normal vocal feeding the main stereo buss and any static effects that are in place; and the other not routed to the stereo buss, but merely used to send signal to the delay.

The fader of this second channel then becomes the send to the delay line, and can be manipulated accordingly. Pushing a fader up to a set position is a lot easier than grabbing an aux send knob each time if you are working manually, and the process can be easily automated if the desk has some automation.

Of course, delay can be applied to the entire vocal from start to finish, making it most apparent at the end of the line after the vocal has stopped (which is the desired effect). Doing this, however, may just clutter up the vocal and obscure it with delay during the singing. The way to get around this is to use ducking.

Ducking can be achieved by feeding the delay outputs through a noise gate set to ducking mode with a split from the dry vocal used to feed the gate's key input. With this method, the delay level will be attenuated when the vocals are present, but will rise to fill the gaps where the vocal has finished.

Careful adjustment of the controls is needed to achieve the most natural-sounding effect.
This technique can work equally well for reverb, and several of the FX units currently available have ducking delay and ducking reverb programs on board which dispense with the need for an external gate.

De-essing
Some recorded vocals can suffer from overemphasised 'S' and 'T' sounds. These sibilants can become especially apparent if delay or bright-sounding reverb is being used on the voice - the T sounds in particular can trigger digital delays to produce repeats that are quite obtrusive in a track. Fortunately sibilants can be tamed by a de-esser, and there are several models currently on the market, available as both dedicated units or as a feature on some voice channel-type units.

Basically a de-esser works like a frequency-conscious compressor, and can be tuned to compress just the frequency where the sibilance occurs (usually somewhere between 4K and 10K, depending on the particular voice). Usually a de-esser will have a control to zoom in on the offending frequency and a control to determine the amount of de-essing carried out, but beware of over-use, as this can sound very unnatural (to the extent that it sounds like the singer is lisping).

In the absence of a dedicated de-esser, any compressor with a sidechain facility can be used. The vocal should be fed through the compressor and an EQ unit patched into the sidechain. If the EQ is set so that the frequencies where sibilance is occurring are boosted, then these frequencies will be compressed.

One method of curbing the effects of sibilance on the reverbs and delays without affecting the dry vocal sound itself is to simply split the vocal and use a de-essed version as an effects send.

Pitch-shifting
A pitch-shifter or harmoniser can be useful for making a vocal sound better, both as a creative effect and in a corrective capacity. A slight detuning of the vocal sound can thicken it up and draw attention away from any small pitching inconsistencies by blurring the edges of the actual pitch by a small amount.

Typical settings on a stereo unit would be a shift up of between 5 and 10ms on one side and a shift down by a similar amount on the other side. Usually there is a delay between the dry and pitch-shifted sounds that can be adjusted to add a bit more thickness. Pitch-shifting of this nature can be applied right through the track, giving a consistent and subtle shift throughout.

For the odd out-of-tune word, a pitch-shifter can be employed to shift it into tune, perhaps with automation being used to switch to the pitch-shifted sound at the appropriate point in the song. Alternatively, a word could be sampled, pitch-shifted, and spun back in.

Finally...
Once the vocal has been EQ'd, panned, effected, and seems to be sitting at the right level in the mix, give the track a good listening to. Turning the speakers down to the minimum can give a good indication of whether the vocal is bedded in nicely, as can listening to the track from outside the door.

Even if the vocal is sitting at the right level through most of the song, there may be sections where it gets buried or becomes too prominent. Fixing this is a job that any automation system can cope with, but when mixing manually using a channel fader, a piece of masking tape next to the fader can be used to mark the fader positions for the various sections.

So, we've come to the end of our quest to record the perfect pop vocal. We said it at the beginning of this series, and now we'll say it again: whether your singer is an ultra-smooth Lisa Stansfield-type, a manic Keith Flint-type, or something in between, it's usually the vocal that determines whether your song will be a hit or a miss. And as a producer, your 'performance' is as important as your singer's is in getting a result.

Over the series, we've touched on every factor you need to consider: from selecting your gear and setting up your studio, through to singers' psychology, comping, backing vocals, and mixing, and no doubt you'll have a few tricks we haven't thought of. Now go forth and make those vocals shine!

Your Analogue Mixer

New equipment won't always solve your production problems, so here's how you maximise the potential of your existing gear...

Some musicians spend an absolute bloody fortune on samplers, MIDI equipment and mics, and still wonder why they are not getting the sound they should be.

One reason for this is that they are not utilising their mixer properly; they are simply using it to balance levels and take the outputs to a mastering medium such as DAT. This is one way to use an analogue mixer - I do this using a small rackmount mixer to submix keyboards, modules and samplers before taking it into the main desk - but this does not utilise the mixer's full range of features.

This article will concentrate on analogue mixers, as there will be a full article on making the most from digital mixers in a forthcoming issue. Even though I thoroughly enjoy mixing on digital desks, I still find recording on analogue desks gives better results. The reasons for this are many, and most are subjective, but I find that when recording down to digital tape (ADAT, DTRS) or hard disk (Fostex D90, Cubase VST, and so on) using an analogue mixer, I can inject some warmth onto a fairly clinical format.

Analogue warmth
What do we mean by 'warmth', though? Ask five different people and you may well receive five different definitions of what warmth is. I see it as the slight distortion that you get on analogue equipment when you drive the levels nice and hard. With analogue tape, this effect is known as tape saturation. Many software manufacturers have produced plug-ins that recreate this effect (Steinberg's Magneto and Red Valve-It are good examples).

The other advantage of recording with analogue desks is that you can make small changes to the sound by adjusting pots and faders in real-time, and it is easy to see at a glance what the settings for each channel are. The advantage of a digital desk at mixdown is that you have your nice fat, warm analogue signals on your digital recording medium and that you are able to set up and save many different variations of mixes and instantly compare them. This is my way of working, and is by no means definitive, but let's look at the features of an analogue mixer and how to maximise them while recording.

Mixer structure
Your mixer has a huge effect on how your recordings will sound, but what makes a good analogue mixer? With the development of very low-noise digital recorders like ADAT, DTRS, and hard disk systems, setting and using your analogue console correctly has become even more important than the type of mixer you are using. There is no point spending £3,000 on a mixer and £1,000 on a mic if you do not know how to set the gain on the console correctly!

Analogue mixers are not as daunting as they first appear. Most of the features we'll be looking at here can be found on a standard 8-buss console such as a Mackie 8 Bus or Behringer MX8000 (a 4-buss or 2-buss desk is pretty similar, but will obviously have less busses).

Normally, at the top of the desk you will have the gain control (see below) along with buttons to select mic/line input and line/tape input (that allows you to select whether a particular channel is monitoring back a mic/line input or a tape return (from your multitrack recorder). You will also usually have your insert points at the top of the desk (see Insert This section on the following page.

Next, the signal goes through the EQ section, which controls the tone of the sound. On simple desks it allows you to cut and boost fixed frequencies, and on more advanced desks you can select the frequencies you are boosting or cutting. Generally speaking, the more advanced the EQ section of your mixer is, the greater control you will have over your sound.

You should really be using a desk with at least a semi-parametric EQ, with the ability to change the frequency of the high and low-mids. The high EQ on less costly desks will usually be fixed around 12kHz, and the low EQ

between 50 and 80 Hz. Full parametric EQ on desks under £1,500 is rare, but you can supplement it by buying a good quality 19" rack EQ unit such as the TLA3012. Put it across an insert, and you have even greater control over the sound you are recording.

Of equal importance is the bypass switch, which enables you to hear what the sound is like before and after any changes you make to the EQ settings. The EQ section of a mixer should be used to fine-tune the frequencies of a sound to help it sit nicely in the mix, rather than to make sweeping changes. If you do not need to EQ a sound, leave it alone.

As a general rule, you should always aim to get the sound you want at source. Get the microphone in the right position to capture your instrument's sound, or select the required synth sound to begin with. If the sound is wrong, don't spend hours with the EQ thinking it will solve the problem, because it won't. Fixing it in the mix is what bad engineers do.

The most hi-tech piece of equipment for telling you how much EQ is required is your ears. Also, don't forget that whenever you boost a frequency you are also boosting the level of that particular channel. This can really screw up your level balance if you're not careful.

Auxiliaries

Below the mixer's EQ section you'll usually find the auxilliaries, which are normally used for adding outboard effects such as reverb, chorus, and delay. They may also be used as additional outputs for recording to tape or for creating separate headphone mixes. My problem is that I always find that I need more auxiliaries than my current desk has, so its definitely a case of 'more is better'.

When using an aux send to add reverb to a channel, you also need a stereo return to hear the effect being applied. Most desks have dedicated aux returns or stereo returns that can be used for effects processors, but on some desks you may need to bring back the aux return on two input channels.

The other reason for wanting to bring it back on input channels rather than a stereo return is that some analogue desks do not have any EQ on the dedicated stereo aux returns. By bringing it back on input channels you can use the EQ to fine-tune the reverb, or even add another effect to the reverb. You can get some great effects by doing this - try it out just for the hell of it.

The auxiliary controls are either designated pre- or post-fade (they may also be switchable between the two). The pre-fade aux is a level control that acts separately from the channel fader. This feeds the signal onto a mixing buss, where it is combined with the pre-fade auxiliary from the other channels. The combined aux signal is then placed under the control of a single aux master control.

Post-fade auxiliaries are very similar to pre-fade, but these are affected by the channel fader settings. This makes them ideal for use as effects sends, because when turning down an instrument using the channel fader, you'd ideally want the effect level to drop as well, unless you are after a particular sort of special effect...

Another use for the auxiliaries is to create a headphone mix for vocalists and other musicians.
Even if you have a very modest studio, the chances are that you will at some point want to work with a vocalist or other live musicians (but watch out for drummers - they will drink your beer fridge dry!) and it can be very frustrating for them to have to record with a crap mix in their cans. More often or not it will also detract from their performance, and then no-one will be happy.

The easiest solution to this problem is to use an aux send to set up a separate level balance for the musician. For example, your control room level might have the harmonic content (the chords) down in the mix, but the vocalist may need it louder to enable to track the pitch more accurately.

A simple way to do this would be to take aux send 3 out to a headphone amp, and then adjust the levels of the channels that the vocalist requires using the individual channels' aux send 3. Using this technique, it is also possible to set up reverb levels as required by the vocalist (most vocalists like to hear some reverb when recording).

Gain setting

I witness many would-be producers setting levels on their analogue mixer using the channel faders, rather than at the gain stage. The signal set at the gain stage determines how clean it is through its path. If the gain is not set correctly, then the preamps will not work efficiently and you could introduce noise or distortion, both of which are difficult to remove once they have entered the signal path.

At the optimum level, distortion and noise become negligible. (why?) The use of visual meters can assist you in this level-setting process, but the most successful technique is to use your ears. When you are setting the input gain on your analogue desk, it is important to PFL (pre-fade listen) the channel you are working on and use the gain pot to help adjust it to the point where there's no distortion/no noise.
I always start with the fader at its normal operating position (0dB on most desks) to ensure I then have the scope to bring the levels up or down as required.

I have often seen many people starting with the fader all the way up to the top of its travel, meaning that it will be virtually impossible to mix properly.

Also, remember to switch out any EQ when setting levels, as cutting or boosting frequencies will affect the levels of a particular channel.

Patchbays
I currently use a digital mixer with an analogue mixer, and have a very extensive patchfield to enable me to easily change routing from one piece of equipment to another.

One of the most noticeable aspects of this patchfield is that the analogue mixer I/O takes up three quarters of the whole thing.

A standard 8-buss mixer has at least 24 channel inputs, 24 group outputs, 24 direct outputs, 24 tape returns, 24 insert send and returns, 8 group outputs, 8 group inserts 6 aux sends, 6 stereo returns, 2 track inputs and 2 track outputs; by anyone's imagination that is, er, quite a few ins and outs.

You have two choices when wiring up your analogue mixer. The first option is to use the 'idiot technique': blow obscene amounts of money on some great pieces of gear and only leave enough money for two pieces of bell wire and some dodgy jacks. Then whenever you need to take the output of one piece of equipment to the input of another you'll have to pull your mixer out to re-patch leads.

The second option is to invest in at least five Behringer Ultrapatch Patchbays (£49). These are the best patchbays I have seen for under £100. You'll also need to spend about £125 for 24-way multicore and mono/stereo jacks from Studio Spares. So for about the cost of a decent GM module, you can have a pro system that eases stress and looks flash.

Mix heaven
If you follow the above guidelines, you will be using your mixer more efficiently, which should result in cleaner, stronger and more professional recordings. Most of us are quite happy to sit down with the manual in order to use our latest sampler (just go with me on this one, allright?) but we just assume we know how our mixer operates. Reading your mixer's manual thoroughly will enlighten you on such subjects as the correct leads to use and the EQ specs, and it will also help you to understand the routing options available.

If you are purchasing a mixer, don't choose one that just about accommodates your current requirements, as you will undoubtedly be adding to your system at some stage. Buy a mixer with as many channels, aux sends, busses and inputs/outputs as possible, a comprehensive EQ section, and preferably, a meter bridge.

Otherwise, you may find yourself having to replace the mixer shortly down the line, and this could end up being more costly than buying a well-specified one in the first place.

A good analogue desk will last years, and will provide a speed, flexibility and control over your sound that no amount of DSP power has yet emulated at an affordable price.

The perfect mix

There are many ways to get your songs to final form. Lets assume, for this article, final form means a beautifully polished piece of music in 16 bit 44.1 khz digital audio (i.e., the "red book" cd audio standard) or a standard wave file. You need to start, of course, with a fully or almost finished song. This is the point where the writing ends and the TweakMeistering begins. I'm going to give you some hard earned tips on Mixing and Mastering.

Mixdown and Mastering, traditionally speaking, are two very separate processes. Mixdown is the art of leveling, equalizing and effecting all the various sources from many tracks down to a stereo Mix. Mastering is the process of taking the stereo mix and putting it in the final album-ready form. Recent software and hardware developments make these processes easier and less expensive than they ever have been in the history of making music. Given that much of the time we can stay in the digital domain we can add processing to our heart's content and maintain a high signal to noise ratio and achieve optimum dynamics for the piece at hand.

TweakHeadz "Overall" Power Mix Parameters

Please consider these parameters not as rules but a starting point for you mixes for the standard pop song or ballad. Of course the instruments change if you are doing techno or symphonies, or ambient stuff, but the reference may still be helpful.

Match the following instruments when soloed in place to the db markers on your mixing desk or your mixdown deck or software.

Set the trims:

Solo each instrument in succession and set the trim so the signal peaks a 0db.

Kick drum 0db +3 eq at 50 Hz +1 db at 3khz -3db 275 hz No FX except maybe subtle ambience. You will tweak the kick again, this is just to get you going.

Snare -2 db eq to taste in the frequencies above 4khz. Add reverb if the song calls for it. Do the best you can to keep it out of the way of the vocal, even if you have to pan it a few degrees.

Lead Vocal 0db use a low cut filter to eliminate rumble and plosive pops around 100-200 hz. Carefully enhance the delicate high end around 15khz to add air and sheen and don't overdo it! This is the trickiest adjustment and may often spell hit or dud. Perfectly center the vocal and pan it not with pan controls, but with very subtle left/right hi freq eq's. Put on the cans (headphones) and make sure its in the absolute center of your forehead.. Every word must be intelligible. Add reverb and delays but don't let it get smeared.

Cymbals -25 db Avoid letting these get in the way of the vocals. Pan them to 2 o'clock and remember their main function is to add the glue to a track to hold the music together--they do not have to be loud or present

Synth pads -20 db Do these in stereo and hard pan left and right with generous effects if needed. However, keep them in the back. Pads indeed are beautiful additions to a song but don't let them overshadow any of the main elements of the song.

Bass -10 db Always front and center. If you use FX restrict yourself to chorusing or a light flange--no reverb.

Rhythm guitar -15 db pan off center eq: use a low cut filter to get rid of any bass and add a mid range eq for a slight narrow boost, but make sure it is not competing with the vocalist's sweet spot.

Percussion -20db- put these elements off center unless they are essential to to basic beat. EQ in a tasteful way if necessary.

Watch the meters when you play the whole mix through the board. You should have peaks at +3db. If what you have is more notch down every fader in 1 db increments until you get there.

Mono Check:

Always check you mix in Mono and look for sudden drop outs or instruments that disappear. That's phase cancellation at work, and it happens with stereo tracks and effects.

No faders above 0db rule:

When getting a mix started follow this religiously. If you find your vocal doesn't sound good unless its at +5db then move everything down 5 db. Conserve headroom. You don't want your mix compromised by that awful crackle at the peak of your song. Now you fine tune to taste. Listen for the quality to "lock". A great mix of a great song will fill you with absolute elation. You'll be blown away and in awe. You will feel in love with it. No kidding. Might sound corny to the less mature among us, but I assure you its true. A great artist friend of mine puts it this way. Greatness in art depends solely on how much love you put in to a work. You put it in, it pays you back, your friends back, and everyone who listens. Moral of this lesson. Never take mixing and mastering lightly. The tiniest fader movements make a difference. Be exacting!

The Mix is a Dynamic, Moving Process

Don't just sit there while your mix goes to tape, or disc, or DAT. If you are using a board, assign the faders to subgroups. For example, if you have 4 subgroups you might want to send your vocal tracks to groups 1 and 2 and everything else to 3 and 4. This way you can slightly alter the balance between the vocalists and the band as the piece goes to tape. This technique, while tricky, can yield outstanding results. You can give the vocalist a touch more edge just when they need that ooomph and when the vocalist takes a break you can subtly boost the band a bit. If you have 8 busses you might dedicate 5 and 6 just to drums and 7 and 8 just to effects, nudging each as is appropriate.

The Role of Compression at Mixdown

On it's way to the recording device, you can patch a compressor/ limiter/gate. The Gate simply cuts out any audio below a certain threshold so that any hiss or noise coming from your synths or mixer is eliminated before the music starts. The limiter keeps your peaks under a certain fixed level and will not let them go higher. A Compressor is a volume slope applied to the audio material goin through it. It can amplify the "valleys" and attenuate the "peaks". Essentially compression reduces the dynamic range we have just struggle to achieve in our mix. You might wonder why you would want that. In many circumstances, you don't want it. However, in the majority of cases you will find it useful, especially if you want your music to be "hot", "have punch" "be as loud as possible", or have the consistency of a radio mix. The stereo compressor also helps balance the song and give it a uniform character we are so used to hearing in commercial music. It essentially gives you the strongest and

smoothest mix and calms down some of the 'jaggged edges' that might disturb the casual listener. However, it is also very easy to make a mix totally lifeless with a compressor and reduce its dynamic power. What started as a powerful orchestral arrangement can end up a wimpy piece of Mall Muzak so be careful and bypass it frequently to make sure you like what you are tweaking up. I think compression works best to attenuate that occasional peak that rips through the roof of a digital audio recorder and ruins the track.

The Role of the Mastering processor

Mastering processors are becoming more popular these days. The TweakMeister likes them. I have noted over and over how the effective use of a mastering processor can transform a good mix into a great master recording. If you have one, you might consider using that in lieu of a compressor at mixdown as mastering processors usually have all the functions and additional functions such as mastering eq, multi-band compression (that is adjustable compression for the bass, mids and highs) as well as limiters and gates. These mastering tools can go a long way to giving your music a unique sonic imprint. There are many uses. In addition to adding the refining touch to your mix as it goes to the recorder, it can be used to give all your songs on an album a consistent uniform character and balance the volume between widely different songs giving your project a professional touch. Using narrow band mid range eqs can give you a very contemporary sounding presence and make your dance tracks come alive with freshness. Pumping the compressor a little at 50-60hz can give you the "kick in the chest" kik drum without wrecking the delicate dynamics of the high end vocals. There are many more applications such as using them to send midi tracks to your digital audio mixer compressed optimally, ducking for voice overs, de-essing, warming through "tape saturation" parameters and Hard Gate effects on individual tracks. Remember Tweakheadz rule of thumb: Any piece of gear can be used in any way as long as it enhances the quality of the final product.

Software Mastering and Post-Production

A good digital audio sequencer will let you master in the digital domain of your computer. Some softwares that I think are of particular merit for mastering are Logic, Cubase, Sound Forge and Vegas. I'm just going to look at Vegas here, because I am enamored with it right now. The main thing is to be able to draw a volume envelope over the whole waveform. Rather than botch a fade 20 times on an analog mixer, simply draw in the perfect fade with the mouse. Where the piece loses intensity, notch it up a tad, to restore your intended dynamism to your mix. Say you have the perfect mix except for one horrible "sp-p-p-lat" where your sequencer choked at bar 72. No prob. Just remix the offending bar again, cut out that piece in Vegas and drop in the new one and let the automatic crossfading give you the absolutely perfect, digitally calculated crossfaded splice. Works! Need to touch up the EQ and do your compression in software? Tweak it in. It's all undoable, so your not going to ruin anything. Decided the mix you did last year really sux? You need to cut out a chorus or fade 5 seconds earlier? Say you did a trance piece but the kick is so wimp that it makes you cringe? Just drag in a looped 808 kik and paint it on the next track, setting the volume and compression to make the whole song whupass. :) Vegas gives you the tools. In fact, I like it better for post pro than as a multi track.

Summing Up:

Whether you are writing industrial hardcore or the darkest ambient, a 100 piece orchestra or a stark minimalist a capella mix, always keep your ears tuned to making an artistic statement, a work of unforgettable beauty. This is the bottom line. The more control your Mixer gives you, the better you can paint the overall image. Working with compressors and mastering processors gives you a shot a polishing that image much like we polish a stone to bring out its colors. Hope this article helped you get a handle on the concepts of the perfect Mix, mastering and post-production

All the Best!

Rich the TweakMeister

MIXING

Without doubt the hardest part of recording is mixing, yet it is also the most enjoyable as this is when everything starts to come together and all the hard work justifies itself. A good mixer paints a picture in sound that attracts the listener and conveys the song clearly and simply. I could sum up a good recording as a series of priorities which are:

The song
The singer
The "feel" or "groove"
The "fiddly bits"

The song is set from the start and a good producer will have chosen a song that has 'something to say' and a good mixer will convey that something to the listener.
The singer is the next most important aspect and a good mixer will allow the singer to be heard and the lyrics to be conveyed clearly but with style. There is nothing more annoying than hearing a track and not being able to distinguish the lyric amongst a babble of instrumentation.
Fortunately you don't hear recordings like that on commercial radio as they just don't get a look in. Engineers are often guilty of cluttering up tracks with all sorts of tricks and garbage that distract from the song and the singer because they know the song so well after days in the studio that they think everyone hears it like they do. If the track is to have a chance of commercial success it must be understandable from the first hearing. Always underestimate the ability of the listener as they are not professional listeners like you.
The "feel" or "groove" is what catches the listeners attention initially and sets up the mood and emotion of the track. This is created by careful balancing of the rhythmic aspects of the track be it drums, percussion or a great guitar groove.
Finally there is the "fiddly bits" as I call them; they are the musical phrases linking lyrics, joining verses to choruses and filling solo sections etc. that are created by the guitar licks, the piano fills, the answering vocal phases etc.

So where to start?

Monitoring Speakers

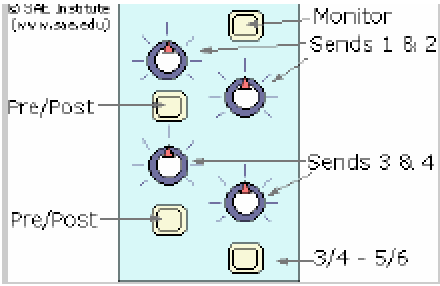
Monitoring speakers come in two types. Nearfield and Main. I like to use both. I work primarily on the nearfield to establish my balances etc. and then every now and then I will switch it up to the big speakers as they give a better idea of the low frequency balance, plus it sounds good eh! (I was a Yamaha NS10 freak for years but now I'm totally sold on the Event 20/20. Well done Event!) To me a big speaker system is like a magnifying glass, it blows the sound up and you can hear more but for a big system to be really good you have to flush mount them and have good speakers and a good amplifier system. Can I say here that I don't like equalised speaker systems. If they don't sound good flat, get another speaker!!

Level Structure

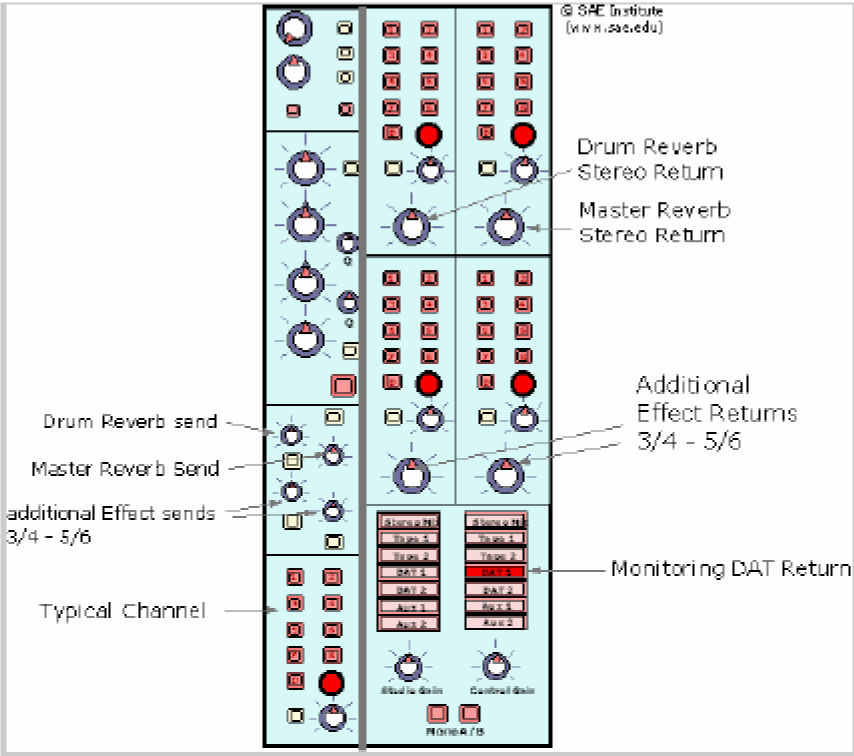
The first important procedure is to setup your console for mixing. The first requirement is to setup your levels to and from your master recorder, usually a DAT. If your console has an oscillator send tone to the DAT and balance left and right channels. Then check that the return to your console, which is what you'll monitor, is balanced correctly left and right. At this stage it is also recommended that you insert your master compressor either in the master stereo output inserts or inline between the console and the DAT and line up correct left/right balance. This procedure is very important as it effects your level structure from then on and if you don't do it now you can end up with your levels all over the shop later.

Aux Sends and Returns

Next you must establish your auxiliary sends and returns.
One of the best ways to get perspective and separation within your mix is to what I refer to as "putting everyone in their own space". You can achieve this through the use of reverb and effects. I like to have one reverb unit dedicated to the drums. No other instruments are sent to this effect, only the drums which will put them in their space. The choice of reverb for drums depends entirely on the track but I start by putting reverb on the snare and going through the presets to



find the one that works best for the track. I find it usually ends up with a bright reverb of shortish length around 1 - 1.2sec reverb time.
Note: A very fine producer in OZ was once quoted as saying "Give me a studio with 10 Midiverbs over a studio with one Lexicon 224XL" We all know what fantastic units the Lexicons are but if it's all you've got you are limited to only one perspective.
Next I'll dedicate a reverb unit to act as my overall reverb effect. I look for the best (not necessarily most expensive) unit in the studio for this will be my master reverb for vocals etc.
In the example above there are 6 sends with 5 & 6 being an option over 3 & 4. I therefore like to use 1 for my drums and 2 for my master verb. Then I can assign the others for effects. I do this so that I can always add master reverb as well as effects if necessary and if I had used say 3, I couldn't put master verb on channels where the effect was assigned to 5. Should I use a stereo or mono send to the effects?? To be perfectly honest I don't think it matters. Most of the stereo input reverb units I find have a mock stereo input, not a true stereo. If you use two sends it really doesn't make a difference unless you are working with the more expensive units like the aforementioned Lexicon, and even then I question the validity of two inputs especially if you are limited in the number of sends.
I then assign the sends 3 - 6 to additional effects like delay, pitch change etc. to act as perspective enhancers. When establishing delays I set them to the track tempo. See Tempo Chart. The idea is to add these perspective effects so you only just hear them when in solo and they appear to disappear when mixed into the track. Bob Clearmountain - the world famous mixer - always had two delays going, one on eighths and the other on 16ths. It puts an air around instruments and if mixed in correctly you won't actually hear them, just sense them. Pitch change is another effect to consider with say the left channel set to -.008 cents and the right to +.008 cents. This effect is great on harmony vocals and it puts them in a different space form the lead vocal. Finally a soft flange or chorus is another effect I'll have as an option for guitars etc. See Effects pages for settings.
Make sure that all your effects are returned through the effect returns and assigned to the master stereo output. If you are fortunate enough to have spare channels on your desk you can return your delay and chorus type effects back through a console channel as this gives you the option of adding master reverb to them and using the channel EQ. Delays can soften if master reverb is added to their returns plus you can attain your feedback from the console instead of using the control on the effect unit. Say you are using send 3 to a delay unit you can feed back to the delay by sending the send 3 on the return back into the unit. N.B. Incidentally, make sure that the dry/ wet or mix controls on your effect units are set to wet as you are only wanting the effect from the units and you won't need any dry sound. (If you are using the Alesis Quadraverb check this as all the default settings have 50% dry and 50% wet.) The returns from effects are usually panned full stereo L/R, but you may wish to bring the drum reverb back half L/R to separate the two.
Your console should now be setup like this



Mixing

Some mixers start with the drums, others start with the vocal. I must admit I start with the drums as they convey the dynamic of a song. Hopefully you will have automation on your console, if not, you must now start setting up a series of moves and remember where and how they occur because, let's face it, the balance within a mix is not static, it varies continuously throughout a song. For example lets say the drummer plays a rimshot snare through the verses and full snare in the chorus. The EQ required on the rimshot snare sound is probably different from the chorus snare sound so I often split the snare return from the recorder into two console channels so I can EQ and effect each separately and automate the switch between the two. For example, the snare in the chorus will probably require more reverb than the rimshot so having a separate channel allows for that. Automation also allows for the tom mikes to be muted when not needed thus reducing the spill of the rest of the kit and cutting out the constant ringing of the toms which occurs with undampened toms. The overhead mikes also will need to be ridden throughout the track, I tend to lower the overhead mikes when the rimshot is playing to achieve a tighter sound, then I lift them in the chorus when the full snare comes in. Reverb on the overheads gives reverb on the cymbals but it also adds reverb to the snare in the chorus and lifts the whole ambient sound of the kit. This has the effect of changing the perspective of the drums in a mix. You can also change the perspective by putting master reverb on the overheads which blends with the drum reverb.

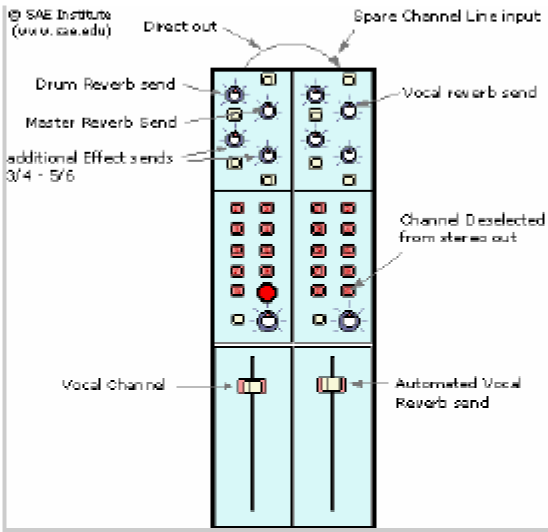
Once we have achieved a reasonable balance of the kit and the dynamics are set in place we can add the bass. The bass and the kick drum will determine the bottom end of the track so the balance between the kick and bass is critical. The kick will give the bass punch and attack when they hit together.

Note: I must say a few words here about bottom end. The big mistake in mixing is to make the bottom end sound too big by adding lots of bottom end EQ to the kick and the bass. You must bear in mind how the track will be played back by the listener. Nowadays everyone has a stereo system with bass boost as an option either as a loudness switch or as a sub bass control. Everyone who has this option has it switched on!! If you get out a few of your favourite recordings and listen to them on your mixing speakers you will find that they are relatively shy in the bottom end and yet when played through your average boom box sound tight and fat. You have to start to understand what a flat response really means and learn to mix that way. If you put a bass on a

VU meter you will notice how much energy there is in the bottom end. A bass peaking to zero will have the same apparent loudness as a highhat peaking to -30db. That's because a hithat has no real bottom end compared with a bass so be careful with your low end EQ on basses and kick drums. I like to solo the two together and EQ them so that they are tight but not boomy.

Add the vocal

OK, so the bass and drums are now at their first mix level so next I will add the vocal and mix it sitting just above the bass and drums. This might mean an EQ change so they all sit tightly together. The vocal might need to be ridden with the automation and I'll probably compress it again to keep the dynamic range within the boundaries of the whole track. I often find that the reverb on the vocal will need to be ridden so that the screaming high notes need more reverb that the quiet intimate sections in the verse. Here I take a feed from the direct out of the vocal channel and bring it up on another channel on the console. I then deselect this channel from the stereo mix output so it goes nowhere but the aux sends are still working. By adding reverb to this channel I can use this channel to ride the reverb on the vocal as an automated send.



Adding the rest

Now we can start to add the fiddly bits like the rhythm guitar and keyboard pads etc. adjusting their balance to fit tightly but not overpowering the vocal. (Please understand I am not defaming guitars etc. by calling them fiddley bits, they are just as important as every other part) The track should now be starting to take shape. If the dynamics of the drums and vocal have been set correctly the placement of the additional instruments will fall into place easily. The vocal harmonies, and solo instruments can now be mixed into the track and we are nearing the completion of the first mixdown.

Note: It is important to keep checking your mix in mono. Unfortunately stereo and mono are not compatible. When you switch to mono, instruments that are panned centre are 3db higher than in stereo so your vocal, kick and snare, for example, will come up in the mix. Some engineers actually make two mixes of a track: One that is full wide stereo with full dynamic range for home listening and one where all the hard left and right signals are panned to the centre or half centre and compressed for radio. It's really hard because if you make a mix sound great on a good home hi-fi it won't have the tightness and punch a mix made for commercial radio will have where the dynamic range is low. It's common practice to make separate mixes of the singles from an album for radio whereas the remaining tracks are mixed totally for home hi-fi. I think you will find that most commercial records are mixed to sound great on FM Radio.

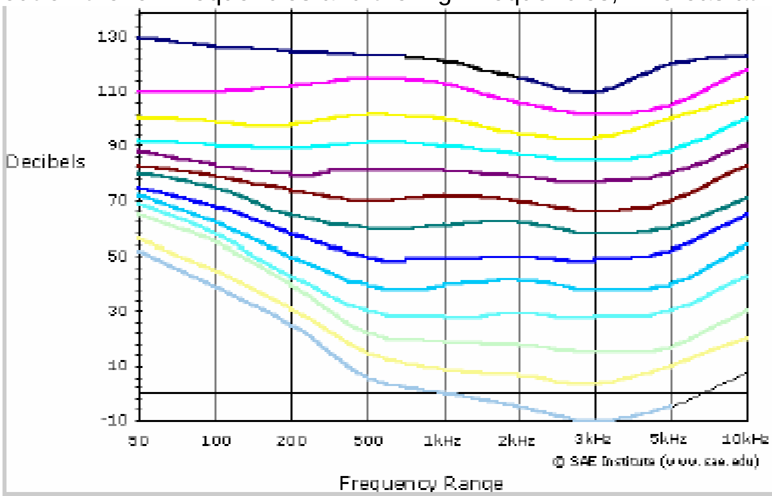
Rest and Recreation

It is important that you constantly give your ears a break during the mixing process as your ears have little compressors in them that will progressively shut your ears down. Have you noticed that when you've been in a loud club with a loud band when you go outside you can't hear as well. It's part of your ears protection system and a cup of coffee in another room watching TV or

something will allow them to start opening back up. I like to " mix from the kitchen" as I call it. This means playing the automated mix and listening to it from an adjacent room with the control room door open, you'd be surprised how clearly you can hear the balance between instruments when you get away from the direct sound from your speakers. The relationship between the bass and kick, the balance within the harmonies, the clarity of the vocals etc. all become clearer when you relax and listen from another room.

Monitoring Level

Unfortunately the human ear is not flat at all levels. Some guys called Fletcher and Munson worked out what the response curve of the ear was and found that at low levels the ear missed out on the low frequencies and the high frequencies, whereas at loud levels it was the opposite.



From the above chart you can see that around 80 - 90db the ear is the flattest. The fact that we don't hear low frequencies and high frequencies at low levels created the Loudness switch on stereo systems which boosts the low and high frequencies to compensate for the ear. Unfortunately, Joe Public doesn't know this but knows that when it is switched in things sound fatter and brighter so they leave it in all the time. It is generally recognised that a level of 85db is where the ear is at it's flattest so don't mix too loud if you want a flat response.

The important thing about mixing is apparent loudness, or relative loudness. If I whisper into a mike and then I shout into a mike the shout will appear louder because I know that shouting is loud. It's the same with mixing. You create an illusion of loudness, everything is relative. You can't get bigger if you are already at your maximum. If I mix a soft acoustic guitar and vocal and peak to zero then bring in a full kit and grunge guitar also peaking to zero it will apparently get louder because I know that drums and guitar are loud. Mixing is the art of making signals that all peak to zero sound as if there is a dynamic range. Nowadays with the excellent compression systems we have most recordings are heavily compressed. I was told of a producer who hired a mixing engineer to mix an album. The guy turned up with racks and racks of compressors and set about compressing every track. He had one compressor for this and another for that etc. In the end the whole mix was pumping away and almost mixed itself. That album went on to sell millions of copies world wide. Those of you who have played with Waves Ultramaximiser will know what compression can do for a mix. If you watch most modern pop recordings on a VU meter the needle is almost static varying only a few db yet the tracks go from quiet intros to full on chorus and solo sections yet still there is only a small variation in level. So setting compression (and limiting) levels is important. I will always have a compressor across the output of my mixes as it helps control the peaks and brings up the loudness of the track but I may use individual compressors on separate channels.

Finally - do take the time to get a good mix. If you don't you have not given justice to all the effort you put into recording it in the first place. It may take a few remixes, so what - it's the final product that counts.